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APPENDIX A
UNIQUE DEPLOYED (TACTICAL)

A.1 INTRODUCTION

This appendix identifies and develops Tactical interoperability requirements as certification criteria for joint networked-communications systems. In pursuing acquisition initiatives, Combatant Commands (COCOMs), military services, and defense agencies shall use this appendix as a guideline for the purchase of commercial off-the-shelf (COTS) equipment as well as for the development of systems that need to interoperate in tactical network environments. The Tactical networked communications community of the Department of Defense (DOD) shall adhere to this appendix in compliance with DOD Instruction (DODI) 8100.04.

This appendix defines unique Deployed (Tactical) requirements for Deployed products and systems. Detailed information and guidance on Requirements Categories, Language, specific terminology, principles, and procedures are provided in the Unified Capabilities (UC) Framework 2013 document.

A.1.1 Purpose

This appendix defines the unique requirements for Deployed products and systems. These are requirements that are not contained in other sections of the Unified Capabilities Requirements (UCR), and define requirements that are modified to support unique tactical users.

This appendix consolidates interoperability certification requirements to the maximum extent possible and incorporates them as part of requirements for the overarching Global Information Grid (GIG) in support of network-centric warfare. This appendix provides guidance for satisfying the certification requirements for Deployed voice systems used as part of an Operational Area Network (OAN), which is the deployed extension of the GIG. This appendix also defines other UCR elements applicable to the Deployed community, and serves as a ready reference to be used by the Joint Interoperability Test Command (JITC) when writing the Deployed annex to the Generic Test Plan (GTP).

A.1.2 Applicability

The requirements described in this appendix apply to Network Elements (NEs), Local Area Networks (LANs) when used in Deployed (Tactical) environments, Deployed Cellular Voice Exchange (DCVX) Systems, and Session Controllers (SCs).

A.1.3 Definitions

Definitions and acronyms are provided in UC Framework 2013, Appendix C, Definitions, Abbreviations and Acronyms, and References.

A.2 CIRCUIT-SWITCHED-BASED DEPLOYABLE NETWORK DESIGNS AND COMPONENTS

Circuit-switched-based deployable requirements defined by previous editions of the UCR remain in effect during the remaining lifecycle of deployed circuit-switched products.

A.3 DEPLOYED VOICE QUALITY

The desired objective for Deployed voice quality is an Mean Opinion Score (MOS) of 4.0 or greater, but it is realized that the network may operate under less than ideal conditions. UC Framework, Appendix A contains additional information on Deployed Voice Quality.

A.4 DEPLOYED NE GENERAL

Section 11, Network Elements, contains the Deployed NE general requirements.

A.5 DCVX SYSTEM

A.5.1 Introduction and Purpose

The following sections describe the requirements that shall be met by all deployed DCVX systems to be certified and used in the OAN tier of the GIG. Requirements are defined at the system level as well as for the various components that make up the cellular system, including protocol requirements. The DCVX is a cellular system with military-unique features (MUFs), and, therefore, is not the same as commercially deployed cellular systems.

It is recognized that not all components are needed for a specific application. The requirements discussed in this appendix are similar to those for a Deployed Voice Exchange-Commercial (DVX-C) and/or SC, and are dependent on the network configuration as well as the specific authorized gateway connection.

A.5.2 Applicability

The requirements within this appendix are applicable to the following:

- All DCVX systems that connect directly or indirectly to the Defense Information Systems Network (DISN) voice systems, including the UC Services Network, Defense Switched Network (DSN), Defense RED Switch Network (DRSN) Secure Phone Gateways, and/or commercial Public Switched Telephone Network (PSTN).
- Procured or leased commercial cellular systems that connect to any DISN service gateway. Commercial cellular services are not currently allowed to be directly connected to DISN service gateways unless the connection is Time Division Multiplexing (TDM) based [e.g., Analog, Primary Rate Interface (PRI), or Integrated Services Digital Network (ISDN)], excluding the use of Signaling System No. 7 (SS7). Future commercial cellular services'

Internet protocol (IP)-based connections will be allowed once the Information Assurance policy and Security Technical Implementation Guidelines (STIGs) are established. In both instances, the DISN service gateway may or may not be protected by a separate or built-in encrypted gateway on the commercial cellular services connection. Encrypted gateway requirements are excluded from the DCVX section.

- Procured or leased cellular systems using leased commercial cellular frequencies that connect to any DISN service gateway.

Terminal devices procured and/or leased, whose primary carrier service is owned and operated solely by a commercial carrier service (e.g., Verizon, Sprint) are not considered elements of a DCVX and are exempt from this appendix. The current version of the UCR is the governing requirements document that takes precedence over the explicit or implicit requirements of subsidiary or reference documents, standards, and specifications. In the event of a conflict, the explicit requirements of the UCR take precedence over the explicit or implicit requirements of any other requirements document except for those requirements specified in the documents listed in Section A.5.3, Policy and Reference Documents.

A.5.3 Policy and Reference Documents

The following policy and instruction documents, in conjunction with the current version of the UCR, will be used as the basis for Approved Products List (APL) certification:

1. Policy for the use of commercial wireless devices, services, and technologies in the DOD GIG, as outlined in DOD Directive (DODD) 8100.2. This directive further promotes joint interoperability using open standards throughout DOD for commercial wireless services, devices, and technological implementations.
2. “Wireless Priority Service (WPS) Industry Requirements for the Full Operating Capability (FOC) for CDMA-Based Systems – Home Location Register (HLR)” or current edition.
3. “Wireless Priority Service (WPS) Industry Requirements for the Full Operating Capability (FOC) for GSM-Based Systems” or current edition.
4. 3G TS 24.067 V3.0.0 (1999-05), 3rd Generation Partnership Project; Technical Specification Group Core Network; enhanced MLPP (eMLPP) – Stage 3 or current edition.

A.5.4 DCVX System Overview

The DCVX systems provide wireless mobile communication services with MUFs and draw their Strategic services by approved DOD authorized gateway switching systems only. The DCVX can be connected to a DVX-C or connected directly to the DSN and/or UC Services Network utilizing UC Session Initiation Protocol (SIP) for TDM and IP switching systems, respectively. The DCVX systems also may be interconnected with other cellular telephone systems, excluding commercial systems, unless the commercial system is procured or leased for DOD usage and is operating in an isolated mode from other commercial provider cellular systems.

When placed in a Deployed environment, the DCVX will have the capability to connect to DSN/UC Services and between other DCVXs and DVX-Cs using UCR-defined protocols such as ISDN PRI, Multilevel Precedence and Preemption (MLPP) PRI (T1.619a), and/or UC SIP. A DCVX system may also be configured to interconnect at the network transmission level with other DCVX systems to provide roaming capability outside the local home base cellular network for supported terminal devices. In support of this roaming capability, the DCVX cellular systems may interconnect based on the interconnection protocol requirements of the appropriate 2G, 3G, and/or 4G standards.

The DCVX terminal devices, often referred to as mobile subscriber cellular handsets, Personal Digital Assistants (PDAs), Smartphones, BlackBerry®s, and any other end user cellular devices, commercial or Government developed, may connect to commercial cellular systems when operating outside the transmission range of the DCVX. Additionally, the cellular terminal devices may have the capability to interface with other wireless networks [e.g., Institute of Electrical and Electronics Engineers (IEEE) 802.11 and IEEE 802.16]. Actual employment of this additional cellular terminal device capability will be by command approval only in the Tactical OAN.

A.5.4.1 DCVX Components

The DCVX is composed of the following three major functional areas: Terminal devices(s), Access Network, and Core Network. Terminal devices can be mobile subscribers' cellular handsets, PDAs, Smartphones, BlackBerry, or any other end user cellular devices, commercial- or Government-developed. With the evolution of cellular technology from 2G to 4G, the primary functional components that compose the DCVX Access and Core Networks are evolving as well. For comparison of the primary functional Access and Core Network components that compose an operational DCVX across the evolutionary changes,

A.5.5 DCVX Operation

The DCVX functions and provides mobile cellular services similar to standard commercial cellular systems with the addition of MUFs. It is based on a two-way cellular radio system that interconnects cell phones with other cell phones and landline stations. When used, the DCVX will provide full mobile cellular coverage in designated deployed environments; this includes training, exercise, and operational missions within COCOM Areas of Responsibility (AORs) or specific geographic areas. User voice, data, and related communications via terminal devices will be similar to landline wired DSN or commercial services. Except for the inherent characteristics of radio transmission, basic service features between the two systems will be similar and transparent to the users. After full mature architectural implementation, the DCVX will function as a wireless adjunct and extension of the joint OAN tier of the GIG. The following configurations, illustrated in Figure A.5-1, DCVX Connection Options, define the operational deployment options of a DCVX.

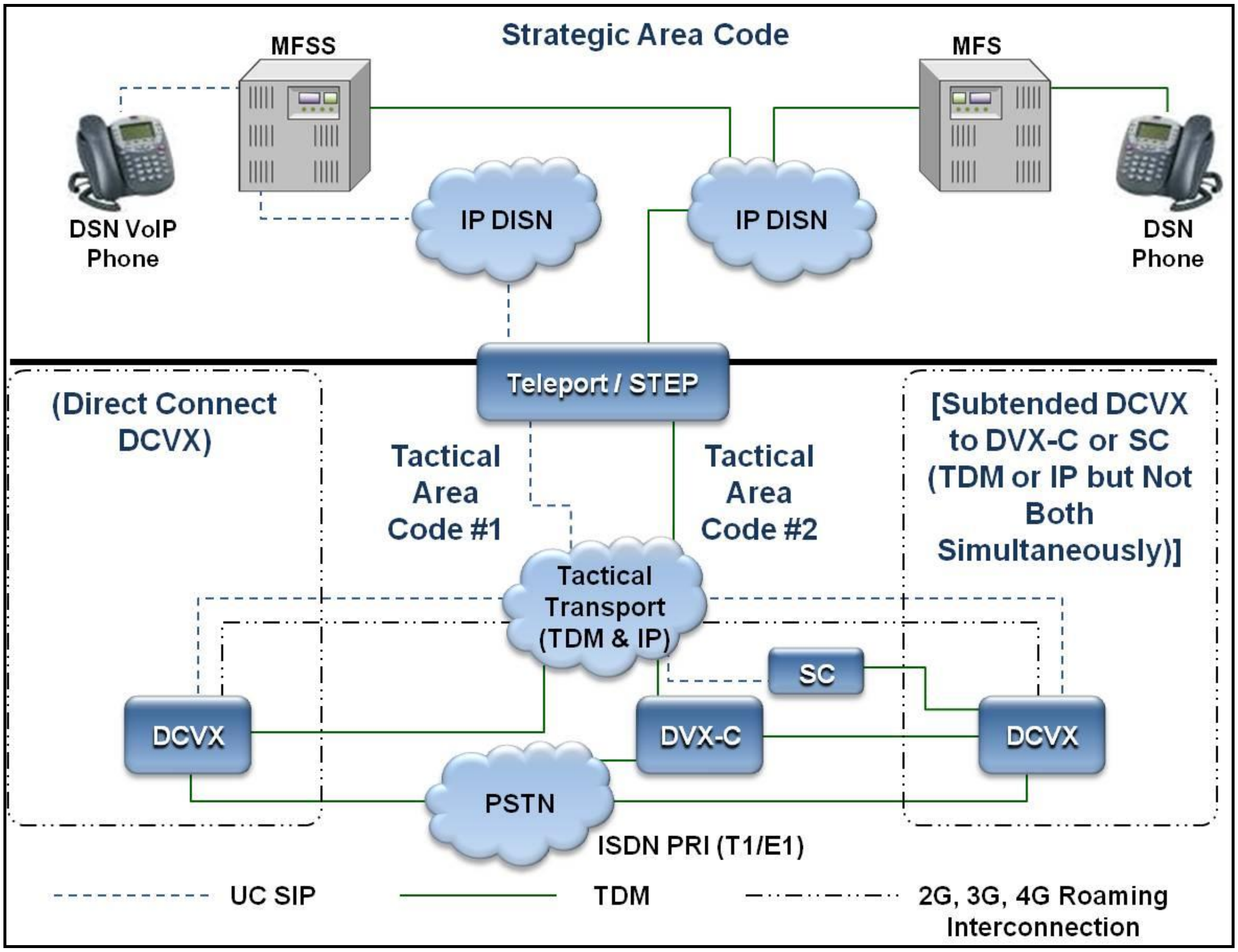


Figure A.5-1. DCVX Connection Options

A.5.5.1 Subtended Deployment Connection

For a subtended deployed connection, the DCVX can reach DSN voice services or UC Services [Voice or Video over IP (VVoIP)] using an existing authorized gateway switch; i.e., DVX-C or a Tactically deployed SC, respectively. To accomplish this, the DCVX can connect to the Tactical TDM and IP transport networks, with one or more of the following interfaces:

- ISDN PRI (T1/E1).
- MLPP ISDN PRI (T1/E1).
- IP UC SIP [Transport Layer Security (TLS) signaling and associated Secure Real-Time Transport Protocol (SRTP) bearer channel].
- IP Non-UC Services (non-Real Time Data, i.e., Best Effort Data).

If the DCVX supports UC SIP in this subtended configuration, connected to a Tactical SC, then the DCVX operates in the Master-Subtended configuration to the Tactical SC. The DCVX can support simultaneous interface connections to the DSN and UC VVoIP/Data networks using

TDM and IP, respectively, but not use TDM and UC SIP protocol simultaneously in support of voice and/or video calls. Current connections to the PSTN and/or other non-Government networks will be limited to ISDN PRI (T1/E1) only. Future IP-based PSTN voice and video service connections will be allowed once Information Assurance policy and STIGs are established.

A.5.5.2 Direct DSN Deployment Connection

For a direct DSN or UC VVoIP connections for UC Services, as well as IP data connections, the DCVX will use the "direct connection" configuration to the Tactical, TDM, and IP transport networks with one or more of the following interfaces:

- ISDN PRI (T1/E1).
- MLPP ISDN PRI (T1/E1).
- IP UC SIP (TLS signaling and associated SRTP bearer channel).
- IP Non-UC Services (non-Real Time Data, i.e., Best Effort Data).

The DCVX can support simultaneous interface connections to the DSN and UC VVoIP/Data networks using TDM and IP respectively, but not use TDM and UC SIP protocol simultaneously in support of voice and/or video calls. Current connections to the PSTN and/or other non-Government networks will be limited to ISDN PRI (T1/E1) only. Future IP-based PSTN voice and video service connections will be allowed once Information Assurance policy and STIGs are established.

A.5.5.3 Networked DCVX Deployment

When a DCVX is deployed in a networked DCVX configuration, a large deployed unit or multiple deployed units within the Tactical OAN may be interconnected with one or more HLR routing tables configured to support cellular terminal device roaming capabilities per the interconnections previously described.

For networked DCVXs within the Tactical OAN in support of a terminal device roaming capability, the DCVX configuration to the Deployed transport network will be with one or more of the following interfaces:

- ISDN PRI (T1/E1).
- MLPP ISDN PRI (T1/E1).
- IP UC SIP (TLS signaling and associated SRTP bearer channel).
- Signaling Transport (SIGTRAN) [Common Channel Signaling 7 (CCS7) over IP].
- 2G, 3G, and/or 4G Standards interconnection protocols transported over DOD Networks.

The extent of terminal device roaming capability will depend on the number and type of interconnections made between the DCVXs within the Tactical OAN and switch lookup routing table updates in the DCVXs themselves.

Current connections to the PSTN and/or other non-Government networks will be limited to ISDN PRI (T1/E1) only. Future IP-based PSTN voice and video service connections will be allowed once Information Assurance policy and STIGs are established.

A.5.5.4 Stand-Alone DCVX Deployment

When a DCVX is used in a stand-alone configuration, the only area served is a deployed unit establishing a Joint Task Force (JTF) and its Command, Control, Communications, and Computers (C4) infrastructure. There is no DSN or PSTN access and no roaming beyond the deployed local network unit cell towers of its area of operation. The DCVX operates solely in an isolated mode.

A.5.6 General Description of Cellular Mobile Features and Technologies

A.5.6.1 Priority Access Service/Wireless Priority Service

Priority Access Service (PAS) provides the logical means for authorized mobile users to queue to the front and obtain priority access to the next available channel in a wireless call path. The goal of the Wireless Priority Service (WPS) is to provide an end-to-end (E2E) OAN-wide wireless priority communications capability to key military personnel during natural or manmade disasters. The WPS is an enhancement to basic cellular service. The full WPS capability can provide priority handling from mobile call origination, through the network, and all the way to the call destination.

The WPS is invoked by keying a special access number (*272) before the destination number on cellular instruments that have been classmarked for the WPS feature. A WPS user may be assigned one of five priority levels (i.e., 1, 2, 3, 4, or 5), with “1” being the highest priority level and “5” being the lowest. Each priority level has user-qualifying criteria that may be tracked for MLPP in DSN or in the UC Network via UC SIP.

When a WPS call is queued for a radio traffic channel from a cellular user and no channel is available, the call is queued according to (1) the highest PAS priority first, and (2) queue entry time (i.e., earliest call first) within the same priority. If the queue for the call sector is full and the caller’s priority is determined to be higher than the level of the lowest priority caller in the queue, then the most recent WPS entry shall be removed, with the new WPS call request queued in accordance with (IAW) items (1) and (2).

A.5.6.2 DOD Global System for Mobile Cellular Band

The current dedicated DOD Global System for Mobile (GSM) band is from 1755 MHz to 1835 MHz, which is a subset of the commercial DCS-1800 band. The remaining Government-owned frequency ranges are 1755 MHz to 1785 MHz for the uplink and 1805 MHz to 1850 MHz for the downlink. There are no non-DOD regulatory challenges associated with the use of the GSM band. The band has been approved for exclusive DOD use and is not authorized for use by any other entity. This band can be used for both voice and data applications to support unique DOD requirements. The Government-owned band may be adjusted in the future, and can be used appropriately at that time.

The band benefits are only effective in a continental United States (CONUS) environment; however, the DOD GSM may be used outside CONUS (OCONUS) with specific host country/countries' authorization. The normal DOD frequency allocation process shall be followed to allow system operation within this band, and Combatant Command/Service/Agency (CC/S/A) planners must ensure that an alternative solution is available before deployment as part of the planning process.

A.5.6.3 Precedence and Preemption

Precedence and preemption can be implemented only in a DOD network. This service has two parts: precedence and preemption. Precedence involves assigning a priority level to a call (wireless or wired). Preemption involves the seizing of a communications channel that is in use by a lower precedence level caller, in the absence of an idle channel. In the DCVX, the Precedence and Preemption capability is Conditional. Precedence and preemption may be provided by enacting enhanced MLPP (eMLPP) or a vendor proprietary version that performs precedence and preemption in the DCVX between the terminal device and the cellular switch. The eMLPP is a cellular version of MLPP and Assured Service in TDM and IP networks respectively. In either version, precedence will be invoked by keying defined digits before dialing the destination number on cellular instruments that have been classmarked for this service. Precedence will function jointly in combination with WPS and will perform E2E as an adjunct to regular MLPP service on the wired DSN and Assured Service on the UC Network. However, in either of the provided versions, if available in the DCVX, eMLPP or vendor proprietary, the connection to the DSN will be MLPP PRI (T1.619a) or use the UC SIP protocol for the UC Network.

Mobile systems, as currently designed, provide a maximum of seven priority levels. The two highest levels (A and B) are reserved for network internal use (e.g., for emergency calls or the network-related service configurations for specific voice broadcast or voice group call services). The second highest level (B) can be used for network internal use or optionally, depending on regional requirements, for subscription. These two levels (A and B) can only be used locally, that is, in the domain of one DCVX. The other five priority levels are offered for subscription and can be applied globally if supported by all related switch elements, and for interworking with ISDN networks providing the MLPP service or Assured Service on UC Network. The seven eMLPP priority levels and their respective mapping to MLPP are defined as follows:

A	Highest, for network internal use	
B	For network internal use or, optionally, for subscription	
0	For subscription:	FLASH-OVERRIDE
1	For subscription:	FLASH
2	For subscription:	IMMEDIATE
3	For subscription:	PRIORITY
4	Lowest, for subscription:	ROUTINE

Levels A and B shall be mapped to level "0" for priority treatment outside of the DCVX area in which they are applied. The vendor-proprietary version will support the five precedence levels as specified for DSN MLPP or UC Assured Service.

A.5.6.4 Code Division Multiple Access Mobile Systems

Mobile Code Division Multiple Access (CDMA) technology uses spread-spectrum telecommunications techniques in which a signal is transmitted in a bandwidth considerably greater than the frequency content of the original information. The latest technology today is based on third generation (3G) that allows high and fast bandwidth, generically called Evolution-Data Optimized (EVDO or EV-DO). This capability supports data usage of the terminal device to allow data connections to DOD networks and future possible use of a Voice over IP (VoIP) softphone on terminal devices when connected to commercial networks for extension of DSN single number presence.

A.5.6.5 GSM Communications Mobile Systems

Early technology for GSM allowed for the use Time Division Multiple Access (TDMA) technology. The TDMA allows several users to share the same frequency. It is the most popular standard for mobile phones in the world. The ubiquity of the GSM standard makes international roaming very common with "roaming agreements" between mobile phone operators. The latest GSM standard is based on an open standard that is developed by the Third Generation Partnership Project (3GPP).

A.5.6.6 4G IMT-Advanced System

Fourth generation (4G) refers to the fourth generation of cellular wireless standards. It is a successor to the 2G and 3G families of standards. The nomenclature of the generations generally refers to a change in the fundamental nature of the service, non-backwards-compatible transmission technology, and new frequency bands. The term 4G refers to an all-IP packet-switched network, mobile ultra-broadband (gigabit speed) access, and multi-carrier transmission. 4G is based on the International Telecommunications Union (ITU)-R standard International Mobile Telecommunications (IMT) Advanced. An IMT-Advanced cellular system must have target peak data rates of up to approximately 100 Mbps for high mobility such as mobile access and up to approximately 1 Gbps for low mobility such as nomadic/local wireless access, according to the ITU requirements. The 3GPP and Worldwide Interoperability for Microwave Access (WiMAX) standards that will meet the ITU IMT-Advanced standard, are the pending 4G-Advanced and 802.16m, respectively. In all suggestions for 4G, the CDMA spread spectrum radio technology used in 3G systems and IS-95, is abandoned and replaced by frequency-domain equalization schemes, for example multi-carrier transmission such as OFDMA. This is combined with Multiple In Multiple Out (MIMO) (i.e., multiple antennas), dynamic channel allocation and channel-dependent scheduling. In the meantime, pre-4G technologies such as first-release 4G Long-Term Evolution (LTE) and Mobile WiMAX, have been available on the market since 2009 and 2006 respectively. However, 4G-LTE does not address the use of voice (i.e., VoIP) at this time. The GSM Association, via the Voice over LTE (VoLTE) initiative, is addressing this omission by selecting a subset of IP Multimedia Subsystem (IMS) standards to deliver E2E voice and Short Message Service (SMS) for LTE devices, including defining roaming and interconnect interfaces. In the meantime, most commercial cellular providers utilize Circuit-Switched Fallback (CSFB), which uses some initial signaling over the LTE Radio Access Network (RAN) and then “falls back” to the 2G/3G TDM RAN to establish the calls.

A.5.6.7 Secure Communications Interoperability Protocol

The Secure Communications Interoperability Protocol (SCIP) is the National Security Agency (NSA)-approved secure voice and data encryption protocol used by DOD, U.S. Government agencies, and civilian authorities. The SCIP is used by the North Atlantic Treaty Organization (NATO) and coalition partners to provide secure voice interoperability between the United States and authorized foreign entities. Application of SCIP is described in detail in Section 3.8, DOD Secure Communications Devices.

A.5.7 DCVX Requirements Terminology

Requirements terminology is defined in UC SIP 2013, Section 1.7, General Requirement Language.

A.5.8 DCVX General

A.5.8.1 Coverage and Signaling Strength

TAC-000010 [**Required**] The signal strength shall not be less than the current GSM (2G, 3G, Pre-4G), CDMA, Mobile WiMAX, and 4G authorized international standards and specifications.

The GSM (2G, 3G, Pre-4G), CDMA, Mobile WiMAX, and 4G technology are spectrum based; therefore, GSM (2G, 3G, Pre-4G), CDMA, Mobile WiMAX, and 4G band, coverage, signal strength, and power are the basis for a planned "area of support." Environment, weather, geography, topography, and adjacent spectrums are elements that must be considered when applying the basis for an area of support. For testing purposes, the generic set of parameters presented in Table A.5-1, Current Cellular Systems Parameters, shall be used for JITC certification either by testing and/or as determined by JITC.

Table A.5-1. Current Cellular Systems Parameters

DCVX GSM/GPRS (2G, 3G, PRE-4G)	
Bands	As provided by standards and/or DOD GSM Cellular Band (e.g., 450 MHz, 850MHz, 900MHz, and 1900 MHz)
Specification on Coverage	As provided by standards (e.g., ITU-R 2G. 2.5G, 3G, 3GSM, UMTS, GSM Edge) (www.itu.int/publications)
Distance Transmit/ Receive	Up to 25 miles depending on topology/manmade structures, and frequencies also determine coverage parameters.
DCVX CDMA	
Bands	As provided by standards (e.g., 450 MHz, 700 MHz, 800 MHz, 850 MHz, 900 MHz, 1700 MHz, 1800 MHz, 1900 MHz, and 2100 MHz)
Specification on coverage	As provided by standards (e.g., TIA, IS-95, 3GPP2, IMT-2000, CDMA 1XRTT, CDMA2000) (www.tiaonline.org)
Distance Transmit/ Receive	Up to 32 miles depending on topology/manmade structures and frequencies also determine coverage parameters.
DCVX (4G IMT-ADVANCED)	
Bands	As provided by standards (e.g., GSM: 700 MHz, 850 MHz, 900 MHz, 1700 MHz, 1800 MHz, 1900 MHz, 2100 MHz and 2600 MHz, Mobile WiMAX: 500 MHz to 3.5 GHz)
Specification on Coverage	As provided by standards (e.g., GSM: 4G-Advanced, Mobile WiMAX, 802.16m, Pre-4G: 802.16-2009)
DCVX (4G IMT-ADVANCED)	
Distance Transmit/ Receive	Up to 25 and 30 miles for 4G-Advanced and WiMAX respectively depending on topology/manmade structures and frequencies also determine coverage parameters
TERMINAL DEVICE	
Bands	As provided by standards (CDMA/GSM/4G-Advanced) and/or DOD GSM Cellular (e.g., 450 MHz, 700 MHz, 800 MHz, 850 MHz, 900 MHz, 1700 MHz, 1800 MHz, 1900 MHz, 2100 MHz, and 2600 MHz, Mobile WiMAX, 500 MHz to 3.5 GHz)
CDMA Specification	As provided by standards (e.g., CDMA (IS95), CDMA2000, CDMA 1XRTT and CDMA 1xEVDO)
GSM Specification	As provided by standards (e.g., GSM (GSM 02.07 Tech. Spec.(ver.7.1.0 Rel. 1998), 2.5G, 3G, 3GSM, GSM Edge)
4G Specifications	As provided by standards (e.g., GSM: 4G-Advanced; Mobile WiMAX; 802.16m, Pre-4G: 802.16-2009)
Distance Transmit/	Up to 8 miles depending on topology/manmade structures and frequencies also determine

Receive	coverage parameters.	
LEGEND		
1xEVDO: One Times EVDO	CDMA2000: Code Division Multiple Access 2000	IMT-2000: International Mobile Telecommunications 2000
1XRTT: One Times Radio Transmission Technology	DCVX: Deployed Cellular Voice Exchange	IS-95: Interim Standard 95
3G: Third Generation	DMSC: Deployed Mobile Switching Center	ITU-R: International Telecommunication Union – Radiocommunication Sector
3GPP2: Third Generation Partnership Project 2	DOD: Department of Defense	
3GSM: Third Global System for Mobile	EVDO: Evolution-Data Optimized	MHz: Megahertz
4G: Fourth Generation	GPRS: General Packet Radio Service	TIA: Telecommunications Industry Association
BSS: Base Station Subsystem	GSM: Global System for Mobile	WCDMA: Wideband CDMA
CDMA: Code Division Multiple Access		WiMAX: Worldwide Interoperability for Microwave Access

A.5.8.2 Protocol/Format

TAC-000020 [**Required**] The DCVX shall support at least one or more of the following protocols:

a. GSM/General Packet Radio Service (GPRS) GPRS [2G, 2.5G, Third Generation (3G), Third Global System for Mobile (3GSM), GSM Edge].
b. Wideband CDMA (WCDMA).
c. CDMA2000.
d. CDMA One Times Radio Transmission Technology (1XRTT).
e. Universal Mobile Telecommunications System (UMTS).
f. EVDO (or EV-DO).
g. Mobile WiMAX (802.16-2009).
h. Fourth Generation IMT-Advanced.
i. 4G-Advanced.
j. Mobile WiMAX Series (802.16m and beyond).

A.5.8.3 MOS and Measuring Methodology

TAC-000030 [**Required**] The DCVX shall support the minimum MOS scores as defined in Section 6, Network Infrastructure End-to-End Performance, or better as measured in any 5-minute interval using ITU-T Recommendation P.862 testing standard. The baseline test environment shall be operated in an open air, clear of obstruction, line-of-sight environment, with the specific requirements as outlined in Table A.5-1. Based on the results, the estimated MOS performance range will be extrapolated and provided in the vendor Letter of Compliance (LOC) based on the Access Network operating at or near full power mode and, at a minimum, operating at a height of 80 feet. The values provided in the vendor letter of compliance (LOC) will be included in the APL report. Refer to Section A.5.14, Submission of Wireless Systems to UC Connection Office (UCCO) for DSN Connection Request, concerning guidelines on submitting the cellular engineering analysis package.

A.5.8.4 Availability

TAC-000040 [**Required**] The DCVX shall have an availability of 99.97 percent, which includes scheduled maintenance.

A.5.8.5 Encryption

TAC-000050 [**Conditional**] Depending upon which of the following encryption types a terminal device provides to support secure calls: SCIP, other NSA-accredited encryption scheme(s), and/or other required accredited encryption schemes as defined in appropriate cellular STIGs, the DCVX must provide appropriate radio and network transport bandwidth to support the terminal device encryption requirements contained in Section A.5.9.4, Terminal Device Encryption.

TAC-000060 [**Conditional**] If a secure call capability is provided in the terminal device(s), then the DCVX shall support SCIP, other NSA-accredited encryption scheme(s), and/or required accredited encryption schemes as defined in the appropriate cellular STIGs. The DCVX that supports SCIP (also known as terminal device) will be required to go secure E2E with another SCIP Phone and/or via a SCIP Gateway if UC SIP is used while the DCVX supports the establishment and maintenance of the secure call.

TAC-000070 [**Optional**] The DCVX may have the capability to provide secure SCIP Gateway functions.

A.5.8.6 Calling Features

A.5.8.6.1 Call Waiting Feature Requirement

The Call Waiting (CW) feature interacts with MLPP and Assured Service for TDM and IP, respectively. If a precedence and preemption capability is available in the DCVX, then the preemption interactions must meet the requirements described in Section A.5.8.10.1, Precedence Call Waiting. Call Waiting is a feature where a line in the talking state is alerted by a CW tone when another call is attempting to complete to that line. A CW tone is only audible to the line with the CW feature activated.

TAC-000080 [**Required**] The CW feature shall generate a CW tone only audible to the line with the CW feature activated.

TAC-000090 [**Required**] The Cancel CW feature is required when CW is active. The user must be able to cancel the CW service. Cancel CW is a feature that allows the user with CW service to inhibit the operation of CW for one call. The user dials the Cancel CW code, obtains recall dial tone, and places a call normally. During this call, the CW service shall be inactive so that anyone calling the CW user shall receive the normal busy treatment, and no CW tones shall interrupt the user's call.

A.5.8.6.2 Three-Way Calling Requirement

The Three-Way Calling (TWC) feature interacts with MLPP and Assured Service for TDM and IP, respectively. If a precedence and preemption capability is provided in the DCVX, then the MLPP interactions must meet the requirements described in Section A.5.8.10.2, Precedence Three-Way Calling (TWC).

TAC-000100 [**Optional**] The TWC feature allows a station in the talking state to add a third party to the call without operator assistance. To add a third party to the call, the TWC customer places the other party on hold, receives recall dial tone, dials the third party's telephone number, and then takes the first line off hold to establish the TWC connection. This may occur at any time after the completion of dialing the second number joining the TWC. After the TWC connection has been established, the customer with the service activated may disconnect the last party added. The customer with the service activated may terminate the TWC call by disconnecting. If either of the other two parties hangs up while the service-activating customer remains off-hook, then the TWC is returned to a two-party connection between the remaining parties.

TAC-000110 [**Optional**] The terminal device may support signaling to allow TWC.

A.5.8.6.3 Conference Calling

The Conference Calling feature is Conditional because it interacts with MLPP and Assured Service for TDM and IP, respectively. If precedence and preemption and conference calling capabilities are provided in the DCVX, then the preemption interactions must meet the requirements described in Section A.5.8.10.3, Precedence Conference Calling.

TAC-000120 [**Optional**] The Conference Calling feature allows the user to establish a conference call involving up to six conferees (including the user). This feature is requested via an access code.

TAC-000130 [**Optional**] The terminal device may support signaling to allow conference calling.

A.5.8.7 Roaming

TAC-000140 [**Optional**] The DCVX system may only support roaming to one or more DCVXs within the Tactical OAN per Section A.5.5.3, Networked DCVX Deployment. The DCVX roaming numbering capability shall support the following:

a. Tactical Global Block Numbering Plan (GBNP).

b. Tactical Routing and Numbering: The DCVX shall be equipped and operationally capable of the dialing format for User Dialing Format to Coalition Forces as defined in NATO Standardization Agreement (STANAG) 4214, “International Rating and Directory for Tactical Communications Systems,” Edition 3, Version T, 7 January 2005, or current edition.

Direct network connections from the DCVX to commercial cellular provider systems in support of terminal device roaming on the commercial cellular provider network(s) are not allowed.

A.5.8.8 Precedence and Preemption

The DCVX may support preemption and precedence under the following conditions:

TAC-000150 [**Optional**] The DCVX may support the cellular version of precedence and preemption, called eMLPP, and/or a proprietary methodology. When precedence and preemption are available, the interface to the DSN/UC Networks and/or the supporting DVX-C shall support one or more of the interfaces as described in Section A.5.11.5, Core Network External Network Trunks and Interfaces.

TAC-000160 [**Conditional**] The DCVX will support a preemption and precedence capability under one or more of the following conditions:

a. The DCVX supports GSM in the DOD GSM cellular band as described in Section A.5.6.2, DOD Global System for Mobile Cellular Band.

b. The DCVX supports the use of leased cellular frequency in one of the bands and protocol(s) listed in Table A.5-1, Current Cellular Systems Parameters.

c. The DCVX supports one or more of the cellular bands and protocol(s), as described in Table A.5-1, Current Cellular Systems Parameters, in an OCONUS environment, where the local Forces-Status Agreement allows eMLPP/proprietary version operation.

d. The DCVX supports one or more of the cellular bands and protocol(s), as described in Table A.5-1, Current Cellular Systems Parameters, dependent on the operational environment and usage of cellular frequencies allowed by local and/or U.S. National Civilian Authorities.

A.5.8.9 Precedence Capability Terminal Device Activation/Deactivation

TAC-000170 [**Conditional**] If a precedence and preemption capability is provided in the DCVX, then the DCVX may be capable of providing on any supported terminal device the user’s Precedence Class Table Assigned features. These features are provided to the terminal device based on the user entering a specified personal identification number (PIN) on the same terminal device. The DCVX will assign to the terminal device the entire user’s precedence capability as defined in the DCVX’s class features table(s). This will allow the user to make precedence calls from terminal devices other than the one assigned or provided to the user. Additionally, the precedence features assigned to that active terminal device can be turned off by reentering the same or different PIN on the terminal device. The precedence capability user's activation or deactivation PIN may be stored in the DCVX or in another database accessible by the DCVX to validate the user's PIN(s) associated with the user's precedence capability. The user's precedence activation or deactivation PIN may be assigned and/or user settable after an initial assigned PIN has been provided.

A.5.8.10Precedence and Preemption Calling Features

TAC-000180 [**Conditional**] If a precedence and preemption capability is provided in the DCVX, then the following applies under the following calling features:

a. If no active call is in progress, then the terminal device will receive precedence notification per Section 2, Table 2.9-1, UC Ringing Tones and Cadences.
b. If a ROUTINE or lower precedence call is in progress to the terminal device, and a calling party calls at a higher precedence level, then the current call will be preempted.

   If a precedence call has been connected to the terminal device and is in progress, then the calling party of equal or lower precedence will receive a notification that the lower precedence call was rejected. The following provides the precedence interactions for calls in progress to terminal devices.

A.5.8.10.1 Precedence Call Waiting

TAC-000190 [**Conditional**] The following Precedence CW treatments shall apply to precedence levels of PRIORITY and above if the precedence and preemption capability is provided in the DCVX.

A.5.8.10.1.1 Busy With Higher Precedence Call

TAC-000200 [**Required**] If the precedence level of the incoming call is lower than the existing precedence call, then precedence CW shall be invoked. In an active call, if the incoming call is PRIORITY precedence or above, the precedence CW tone shall be applied to the called party per UC SIP 2013, Section 6, Table 6.1-4, UC Information Signals.

A.5.8.10.1.2 Busy With Equal Precedence Call

TAC-000210 [**Required**] The DCVX shall provide the precedence CW signal to the called station per UC SIP 2013, Section 6, Table 6.1-4, UC Information Signals. The DCVX shall apply this signal regardless of other programmed features, such as call forwarding on busy or caller ID. The called station shall be able to place the current active call on hold, or disconnect the current active call and answer the incoming call.

A.5.8.10.1.3 Busy With Lower Precedence Call

TAC-000220 [**Required**] The DCVX shall preempt the active call. The active busy station shall receive continuous preemption tone until an on-hook signal is received and the other party shall receive preemption tone for a minimum of 3 seconds. After the current call is terminated and the terminal device is idle, the station to which the precedence call is directed shall be provided precedence notification ring per Section 2, Table 2.9-1, UC Ringing Tones and Cadences, or comparable vibration cadence. The station shall be connected to the preempting call after going off-hook.

A.5.8.10.1.4 No Answer

TAC-000230 [**Required**] If, after receiving the precedence CW signal, the busy called station does not answer the incoming DSN call within the maximum programmed time interval, the switch shall treat the call IAW Section 2.2.10, Precedence Call Diversion.

A.5.8.10.2 Precedence Three-Way Calling (TWC)

TAC-000240 [**Conditional**] If precedence and preemption and TWC are provided in the DCVX, then the following TWC requirements apply:

a. [**Required**] In TWC, each call shall have its own precedence level. When a TWC is established, each connection shall maintain its assigned precedence level. Each connection of a call resulting from a split operation shall maintain the precedence level that it was assigned upon being added to the TWC.

b. [**Required**] The DCVX shall class mark the originator of the TWC at the highest precedence level of the two segments of the call. Incoming calls to lines participating in the TWC that have a higher precedence than the higher of the two segments shall preempt unless the call is marked non-preemptable.

c. [**Required**] When a higher precedence call is placed to any one of the TWC participants, that participant receives the preemption tone per UC SIP 2013, Section 6, Table 6.1-4, UC Information Signals. The other two parties shall receive a conference disconnect tone. This tone indicates to the other parties that one of the other TWC participants is being preempted.

d. [**Required**] In a TWC call where each connection is established at a different precedence level, the precedence level of the participant who initiated the TWC call shall be assigned the highest precedence of the two connections.

A.5.8.10.3 Precedence Conference Calling

TAC-000250 [**Conditional**] If precedence and preemption and conference calling are provided in the DCVX, then the following precedence conference calling requirement is required:

a. [**Required**] All addresses shall be processed at a precedence level equal to that precedence level dialed by the conference originator.

(1) If all conference bridges are busy, then ROUTINE precedence conference call attempts shall be connected to a "line busy" tone per Table A.6-2, CVVoIP Information Signals, and call attempts at precedence levels above the ROUTINE precedence shall re-examine all conference bridges on a preemptive basis.

(2) A conference bridge that is busy at the lowest level of precedence stored for all units shall be preempted for a higher precedence conference call.

(3) When a conference bridge is preempted, a 2-second burst of preemption tone per UC SIP 2013, Section 6, Table 6.1-4, UC Information Signals, shall be provided to the conferees on the existing conference. The existing connections to the bridge shall be dropped, and the bridge shall send an on-hook signal automatically to the associated switch ports to permit the new connections to be established.

(4) Where the requesting precedence level is equal to or lower than the existing conference, the connection shall be denied, and the caller shall be provided a Blocked Precedence Announcement (BPA) per Section 2.9.1.2.2, Announcements.

A.5.8.10.4 Voice Mail

The Voice Mail feature interacts with precedence and preemption. If precedence and preemption capability and voice mail are provided in the DCVX or voice mail added externally, then the precedence and preemption interactions must meet the requirements described in Section 2.25.2.3, Precedence Call Diversion.

TAC-000260 [**Optional**] The DCVX may provide ROUTINE calls only voice mail capability for users. Additional features, such as message forwarding, may be provided in addition to a basic voice mail capability provided they do not interfere with precedence and preemption if the capability is provided in the switch.

A.5.8.10.4.1 Precedence and Preemption Interaction With Voice Mail

TAC-000270 [**Conditional**] If precedence and preemption are provided in the DCVX and voice mail capability is provided internally to the DCVX or connected externally to the DCVX as an adjunct, then the following requirement applies:

a. [**Required**] The DCVX shall divert all precedence calls above ROUTINE that are destined for voice mail IAW Section 2.25.2.3, Precedence Call Diversion.

A.5.8.11Management Capabilities for Terminal Devices

TAC-000280 [**Required**] The DCVX shall have the capability to manage its supported terminal devices as published in its users' database [e.g., HLR or Mobility Management Entity (MME)] so it can assign, transfer, or terminate services, features, and calling capability to include telephone numbers for its terminal devices.

A.5.8.12 Security

TAC-000290 [**Required**] All components of the DCVX shall meet security requirements as outlined in DODI 8510.01 and the applicable STIG(s).

A.5.9 Terminal Device-Specific

Cellular handsets, often referred to as mobile subscribers, handsets, PDAs, Smartphones, BlackBerrys, and any other end user cellular devices, commercial- or Government-developed, are herein referred to as terminal devices. The terminal device is the interface between the user and the cell network. The terminal device can be a handheld unit, a mounted mobile device, or a fixed location device.

A.5.9.1 Terminal Device

TAC-000300 [**Required**] The terminal device shall provide the following status information to the network:

a. Powered on.
b. Moved to a new location.
c. Alerting.
d. Dialing.

TAC-000310 [**Required**] The terminal device shall display the following status information to the end user:

a. Signal strength.
b. Battery capacity.
c. Roaming status.
d. Service not available.
e. Call progress status.

TAC-000320 [**Required**] If no STIG exists for the terminal device, then the terminal device shall have the ability to provide key-locking ability to lock the terminal device’s keypad and unlock the keypad after providing the appropriate key sequence or PIN entries as provided by the vendor in the terminal device. The lock and unlock key sequence or PIN shall be set by the user. If the user PIN is unavailable or not supplied, then an administrator method, which can be vendor proprietary, shall unlock the terminal device.

TAC-000330 [**Optional**] The terminal device may have the capability to support WPS on commercial networks and/or DOD networks where provided when not connected to and functioning on a DOD precedence and preemption network.

TAC-000340 [**Conditional**] Removable and Exchangeable Subscriber Identity Module (SIM): If a SIM card is utilized, then the SIM card in commercially available terminal devices shall be removable and exchangeable into other similar commercially available terminal devices that are compatible with the DCVX system (applicable to a GSM-based system). This excludes secure terminal devices and other terminal devices not readily commercially available.

A.5.9.2 Terminal Device Signaling

TAC-000350 [**Required**] The terminal device shall provide information to allow the DCVX to identify the terminal device when the terminal device is powered up, successfully registered, and in active call status.

A.5.9.3 Terminal Device Frequency Band Support

A terminal device that supports more than one frequency band has a high connection and reliability capacity.

TAC-000360 [**Optional**] A terminal device may support multiple (e.g., five) frequency bands as specified in Table A.5-1, Current Cellular Systems Parameters, for each protocol supported in Section A.5.8.2, Protocol/Format.

TAC-000370 [**Optional**] The terminal device may also support roaming and interconnecting with commercial cellular networks when operating outside the transmission range of the home based DCVX and other supporting DCVXs interconnected in support of roaming within the Tactical OAN.

A.5.9.4 Terminal Device Encryption

TAC-000380 [**Conditional**] If SCIP and/or other NSA-accredited encryption are implemented in the terminal device, then the SCIP and/or other NSA-accredited encryption-capable terminal device shall have the capability to go secure to provide E2E encryption to another secure cellular-capable terminal device, and via the DCVX, to a non-cellular NSA encryption-capable device per the requirements specified in Section 3.8, DOD Secure Communications Devices (DSCD). The SCIP and/or other NSA-accredited encryption device shall provide E2E encryption within the DCVX, from DCVX to DCVX (roaming) and from DCVX to external networks such as DSN, UC Network, and/or PSTN.

TAC-000390 [**Optional**] The terminal device may support other non-NSA encryption schemas, such as Advanced Encryption Standard (AES) encryption as used by the Government Emergency Telecommunications Service (GETS) system.

A.5.9.5 Device Battery

TAC-000400 [**Required**] The commercially available nonsecure terminal device that is readily available must have a battery that shall provide as a minimum 6 days standby time in total and 3 hours nonsecure talk time in total but not both requirements sequentially on the same battery charge. The NSA encryption secure terminal devices [e.g., PDA Secure Mobile Environment Portable Electronic Device (SME PED)] must provide their specified battery and secure or nonsecure talk time. All other terminal devices must provide their specified battery and nonsecure talk time and/or secure talk time, if applicable.

TAC-000410 [**Required**] The terminal device shall have the capability, when the primary battery is removed or drained, to retain primary network and user settings on the device before another primary battery is installed or recharged. This is required to ensure the terminal device is able to reconnect to the DCVX upon power-up.

A.5.9.6 Terminal Device Secure Call Handling

TAC-000420 [**Conditional**] If the terminal device supports SCIP or other NSA-accredited encryption scheme(s), then the terminal device and/or DCVX system will provide classified secure call handling features, as defined in Section 11.3.8, Secure Call Handling, if conversion is made from TDM to IP Network boundaries.

A.5.9.7 Terminal Device Display and Alerting Features

The terminal device shall have the following display and alerting features:

TAC-000430 [**Required**] Power-On Status. When the terminal device is powered on, the display shall indicate:

a. Signal strength.
b. Remaining battery capacity.
c. Active call status.
d. Registration results (either success or failure).

TAC-000440 [**Required**] ROUTINE Call Alerting. The idle, registered terminal device shall provide or be provided with an auditory and/or visual display alert for incoming ROUTINE calls.

TAC-000450 [**Optional**] Precedence Call Alerting. The DCVX may be required to meet the eMLPP functionalities specified in Section A.5.6.3, Precedence and Preemption. The eMLPP references or uses a proprietary methodology. If precedence and preemption capability is provided, then, upon receiving a precedence call, the idle, registered terminal device will provide or be provided with a precedence alert and/or tone notification. Whether using eMLPP or a proprietary version, the terminal device shall issue the same alerting tone(s) for precedence calls

IAW eMLPP requirements. Upon notification, the user will have the capability to select or reject the call of higher precedence.

A.5.10 Access Network-Specific

Specific Access Network capability is as follows.

A.5.10.1Signaling

TAC-000460 [**Required**] The Access Network will determine which channel to use for call setup IAW the appropriate supported protocols listed in Section A.5.8.2, Protocol/Format, as outlined in Table A.5-1, Current Cellular Systems Parameters.

A.5.10.2Strength

TAC-000470 [**Required**] The Access Network will monitor the terminal device for signal strength and transfer the terminal device to the stronger cell when necessary IAW the appropriate supported protocols listed in Section A.5.8.2, Protocol/Format.

A.5.10.3Protocol/Format

TAC-000480 [**Required**] The Access Network shall support one or more of the protocols listed in the DCVX general requirements, in Section A.5.8.2, Protocol/Format, and as outlined in Table A.5-1, Current Cellular Systems Parameters.

A.5.10.4Coverage

TAC-000490 [**Required**] The Access Network will assign the strongest cell to the terminal device per the standards. The coverage area this system will provide shall be IAW the GSM (2G, 3G, Pre-4G), CDMA, Mobile WiMAX and/or 4G standards and specifications IAW Table A.5-1, Current Cellular Systems Parameters, and in Section A.5.8.2, Protocol/Format. Actual coverage will depend on topology and/or manmade structures and frequencies.

A.5.10.5Preemption

TAC-000500 [**Conditional**] If precedence and preemption capability is provided in the DCVX, then, in the event of a preemption for reuse, the Access Network and/or Core Network must disable the old call and maintain the current channel assignment to the terminal device in order to allow the set up of the new call. In the event where there are no idle channels available and preemption for reuse does not occur, then when a precedence call is received, the DCVX will find the lowest precedence channel and preempt that channel to allow for the higher-level precedence call to be completed.

A.5.11 Core Network-Specific

Because of the differences between the various cellular generations (2G, 3G, Pre-4G, 4G), it is not feasible to identify specific component requirements. Thus, this appendix refers to Core Network functionality instead. Additionally, the HLR functionality is not required to be a local component part of the Core Network, but it will be necessary for the Core Network to access a home location register at some location to determine the attributes of its supported terminal device. Whether the home location registry functionality is local with the Core Network or it is remotely queried, the home location registry functionality is a component of the DCVX under test.

A.5.11.1 Visitor Location Register Functionality

TAC-000510 [**Required**] The Core Network shall maintain a Visitor Location Register Functionality to allow service to any authorized active terminal device within its domain per in Section A.5.8.2, Protocol/Format. Visitor Location Register (VLR) functionality may be updated by the DCVX resident HLR functionality, a shared HLR functionality with another DCVX, and/or via roaming between DCVXs.

A.5.11.2 Home Location Register Functionality

TAC-000520 [**Required**] The Core Network shall connect to an HLR functionality to determine the attributes of the terminal device currently being served by the DCVX. The HLR Functionality can be co-located with the Core Network or accessed remotely. Access to the remote HLR Functionality may be by one or more of the following connection types:

a. ISDN PRI (T1/E1).
b. MLPP ISDN PRI (T1/E1).
c. IP UC SIP (signaling and associated bearer channel).
d. SIGTRAN (CCS7 over IP).
e. 2G, 3G, and/or 4G Standards interconnection protocols transported across DOD Networks.

TAC-000530 [**Required**] HLR Storage. The HLR Functionality must store and support information on each terminal device registered to the network that the HLR Functionality serves.

TAC-000540 [**Required**] HLR Change and Propagation. The HLR Functionality must support changes to the terminal device information. Once the HLR receives the supported change information, the HLR, whether local or remote from the Core Network, has 3 minutes to propagate the change information to the VLR Functionality. If the DCVX supports roaming, then the HLR change must also propagate to the querying VLRs.

TAC-000550 [**Conditional**] Intra-DCVX Queries. If a roaming capability is supported in the DCVX, then the HLR Functionality must support queries from other DCVXs using specified protocol methods for obtaining terminal device information [e.g., GSM (2G, 3G, Pre-4G), CDMA, Mobile WiMAX, and/or 4G standards] based queries.

A.5.11.3 Equipment Identity Register Functionality

TAC-000560 [**Required**] To validate terminal devices to prevent a compromised terminal device from connecting to the cellular switch and obtain services, an Equipment Identity Register (EIR) functionality must be provided and integrated to work in conjunction with the Terminal Device Authentication Center functionality as stated in Section A.5.11.4, Terminal Device Authentication Center Functionality, to prevent compromising the DCVX.

A.5.11.4 Terminal Device Authentication Center Functionality

TAC-000570 [**Required**] To authenticate terminal devices as valid terminal devices associated with the DCVX, the cellular switch will use standard cellular techniques, industry best practices, and/or vendor proprietary processes integrated into the switch.

TAC-000580 [**Optional**] Terminal devices not assigned to the supporting DMSC HLR (e.g., roaming terminal devices) may be supported for authentication via the industry standard(s) and/or industry best practices for roaming authentication.

A.5.11.5 Core Network External Network Trunks and Interfaces

TAC-000590 [**Required**] The Core Network shall support one or more of the following TDM and/or IP trunks and interfaces. The Core Network can support simultaneous interface connections to the DSN and UC VVoIP/Data networks using TDM and IP respectively, but not use TDM and UC SIP protocol simultaneously in support of voice and/or video calls.

A.5.11.5.1 TDM Support

TAC-000600 [**Conditional**] If TDM trunks are supported, then the following requirements apply as directed:

a. [**Required**] The Core Network will support ISDN PRI (T1/E1) as defined in Section 2.25.1, National ISDN 1/2 Basic Access for trunks that connect to the DSN/PSTN without MLPP capability.

b. [**Conditional**] If a precedence and preemption capability is provided in the DCVX, then the Core Network will support MLPP PRI [American National Standards Institute (ANSI) T1.619a, ITU Q.955.3 and/or Q.735.3] per Section 2.25.2.7, ISDN MLPP PRI.

c. [**Conditional**] The Core Network may support a DS1 Interface (e.g. PCM-24, PCM-30) per Section 11.2.3.4, DS1 Interface.

A.5.11.5.2 UC SIP IP Trunking Support

TAC-000610 [**Conditional**] If UC SIP IP trunks are supported, then the DCVX shall comply with the stated requirements of an SC, and if required, act as a SIP Back-to-Back User Agent (B2BUA). The Core Network and terminal devices supporting UC VVoIP Services are required to meet the conditions as stated in Section 8, Information Security.

A.5.11.5.3 DCVX Interconnection (Roaming)

Including the connections provided in Section A.5.11.5.1, TDM Support, and Section A.5.11.5.2, UC SIP IP Trunking Support, one or more of the following connections can be used for connecting DCVXs together on DOD networks within the Tactical OAN in support of roaming capability and/or querying the local or remote HLR Functionality. Neither connection type below shall connect to the PSTN and/or other non-Government networks.

TAC-000620 [**Optional**] SIGTRAN: The Core Network may support CCS7 over IP using SIGTRAN IAW Internet Engineering Task Force (IETF) Request for Change (RFC) 2719, and other associated supporting RFCs.

TAC-000630 [**Optional**] 2G, 3G, and/or 4G Standards: The interconnection portion of the protocols contained within the 2G, 3G, Pre-4G, Wideband WiMAX, and/or 4G Standards, as delineated in Section A.5.8.2, Protocol/Format, may be used to interconnect DCVX systems when said protocols are transported over DOD operated and/or controlled networks.

A.5.11.5.4 Non-MLPP Networks Support

TAC-000640 [**Optional**] The Core Network may support an ISDN PRI (T1/E1) non-MLPP trunk for connecting to the PSTN and/or other non-Government networks. ISDN PRI (T1/E1) requirements are contained within Section 2.25.1, National ISDN 1/2 Basic Access.

A.5.11.6Call Handling

TAC-000650 [**Required**] The Core Network shall handle both intraswitch calls and calls to and from the DSN, PSTN, and/or UC Services Network, while recognizing a powered-on terminal device that comes into its operational area.

A.5.12 Security

TAC-000660 [**Required**] All components of the DCVX shall meet security requirements as outlined in DODI 8510.01 and the applicable STIG.

A.5.13 DCVX Network Management

TAC-000670 [**Required**] The DCVX is to be managed by at least one or more of the following:

a. [**Optional**] A front or back panel and/or external console control capability shall be provided for local management.

b. [**Optional**] Remote monitoring and management by the Advanced DSN Integrated Management Support System (ADIMSS) or similar Network Management (NM) systems developed by DOD Components. The following requirements apply:

(1) [**Required**] Data Interface: The NE shall provide NM data/monitoring via one or more of the following physical interfaces:

(a) Ethernet/Transmission Control Protocol (TCP)/IP (IEEE 802.3).

(b) Serial (RS-232)/Asynchronous.

(c) Serial/Synchronous (X.25 and/or BX.25 variant).

All data that is collected shall be accessible through these interfaces. For NM purposes, the NE must provide no less than two separate data channels. They may be physically separate (e.g., two distinct physical interface points) or logically separate (e.g., two user sessions through a single Ethernet interface). The data may be sent in ASCII, binary, or hexadecimal data or ASCII text designed for screen/printer display.

The data channels shall be used for and, as such, must be capable of providing:

i. Alarm/Log Data.

ii. Accounting data [e.g., Call Detail Record (CDR)].

iii. Performance Data (e.g., traffic data).

iv. DCVX access (to perform DCVX data fill administration and network controls).

(2) [**Required**] Fault Management: The DCVX shall detect fault (alarm) conditions and generate alarm notifications. The alarm messages must be sent to the assigned NM Alarm channel in near-real time. No alarm restriction/filtering is necessary. In addition to the data formats in Section 11.2.4, Device Management, alarms may be sent as Simple Network Management Protocol (SNMP) traps. If this channel is also used to output switch administrative log information, then the alarm messages must be distinguishable from an administrative log message.

(3) [**Required**] Configuration Management: Requirements for this feature shall be in accordance with Telcordia Technologies GR-472-CORE, Section 4.

A.5.14 Submission of Wireless Systems to UCCO for DSN Connection Request

TAC-000680 [**Required**] The DCVX systems shall be engineered so that the Access and Core Networks achieve the required performance requirements in their specific deployed environment.

The user shall submit a network design and engineering performance analysis with supporting calculations to meet minimum MOS performance with the request for DSN, PSTN, and/or UC Services Network connection. For certification procedures, the UCCO submittal shall include wireless security compliancy as identified in Section A.5.12, Security.

A.6 DEPLOYED TACTICAL RADIO

The requirements discussed in this appendix refer to post-2012 system deployments. This appendix does not discuss transition between current system deployments and the systems described herein.

A.6.1 Introduction and Purpose

The following sections describe the requirements that shall be met by all deployed Tactical Radio Networks (TRNs) for them to be certified and used in the OAN tier of the GIG. Requirements are defined at the system level, as well as the various components that make up the radio networks, including protocol requirements. Several of these requirements reflect changes described elsewhere in this UCR. These will be indicated in the text.

The scope of this appendix is limited to push-to-talk (PTT) TRNs. Future updates will address radios that have the ability to dial directly to a DISN VoIP End Instrument (EI).

A.6.2 Applicability

The requirements within this appendix are applicable to all PTT-based TRNs that connect directly or indirectly to the DISN VoIP services.

The current version of the UCR is the governing requirements document that takes precedence over the explicit or implicit requirements of subsidiary or reference documents, standards, and specifications, except for those requirements specified in the documents listed in Section A.5.3, Policy and Reference Documents.

A.6.3 Policy and Reference Documents

The policy and instruction documents in Section A.5.3, Policy and Reference Documents, will in conjunction with the UCR be used as a basis for APL certification.

A.6.4 TRN System Overview

The TRNs provide wireless communication services with MUFs. They differ from commercial, standards-based cellular networks in that individual radios within the TRN can communicate with each other, without the need for a base station, Base Station Controller (BSC), or constituent signaling, or interconnect equipment. The TRNs use multicast-based Radio Frequency (RF) transmissions to enable a set of radios using the same frequencies to communicate with each other.

The lower portion of Figure A.6-1, TRN Connectivity, shows the architecture for a notional TRN and how the TRN connects to the DISN UC-compliant service. This connection is facilitated by a new UCR function called the Radio Bridge Function (RBF). The RBF is a component of a deployed SC, or a component of a radio within the TRN.

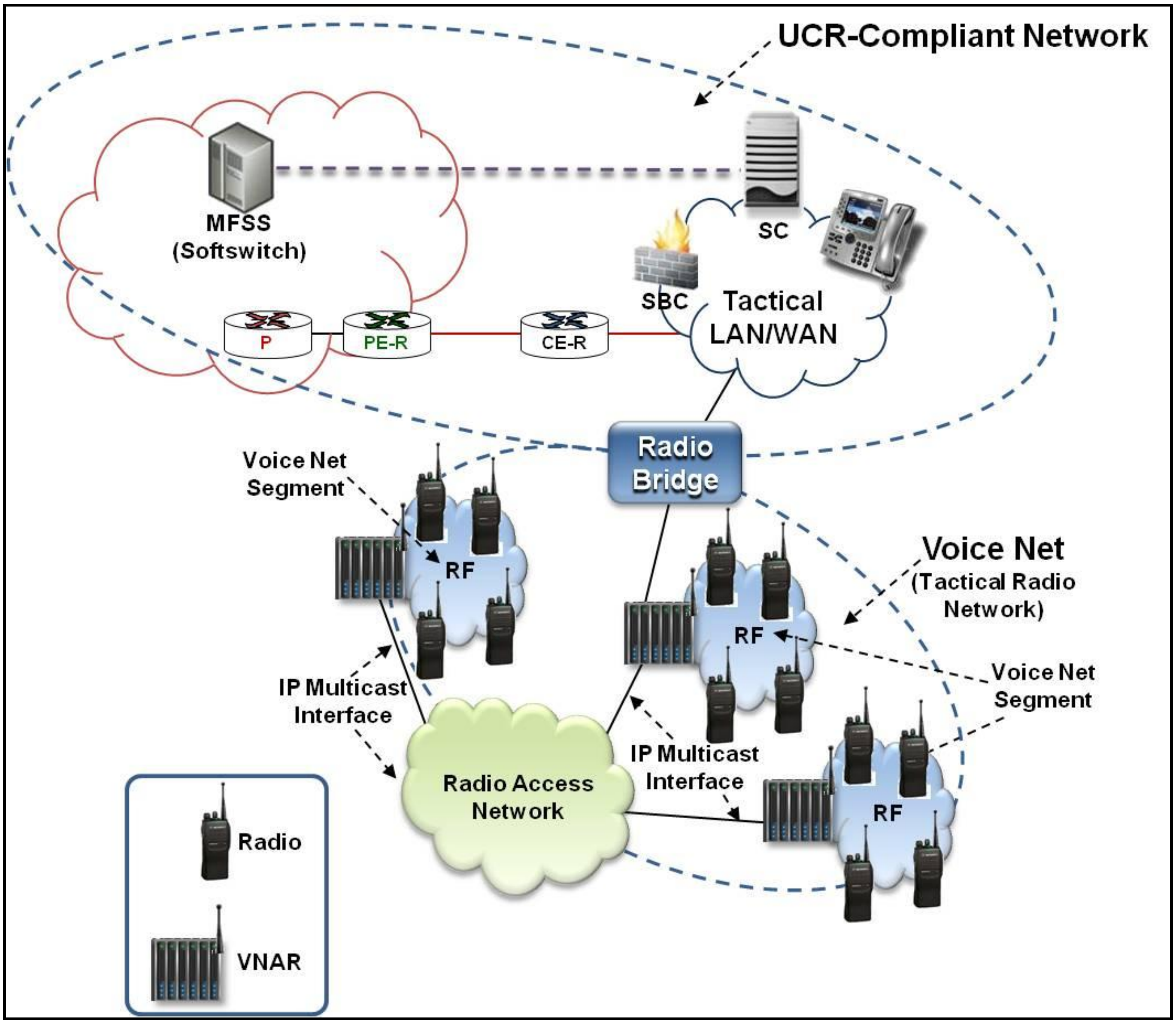


Figure A.6-1. TRN Connectivity

The upper portion of Figure A.6-1 shows elements of the UC-compliant network and interconnects, as described elsewhere in this UCR.

The TRN, also known as a "Voice Net" for this description, is composed of Voice Net Segments. Each Voice Net Segment is a group of radios, which communicate on a common set of frequencies. At least one radio in each Voice Net Segment is designated as a Voice Net Access Radio (VNAR).

TAC-000690 [**Conditional**] A VNAR performs as many as three roles, depending on the type of Voice Net. It acts as a conventional radio to communicate with other radios in its Voice Net Segment. If there is more than one Voice Net Segment in a Voice Net, then the VNAR shall communicate with VNARs in the other Voice Net Segments using, what could be, a proprietary, packet-based RAN. At least one VNAR in a Voice Net also may act as a UCR-compliant (e.g., APL-listed) EI, to enable voice communications between the Voice Net and UCR-compliant VoIP EIs.

The methods and formats for VNAR communications over the RF links and the RAN depend on the type of technology used to create the Voice Net. Some Voice Nets operate in multicast PTT mode, where one party speaks and the others listen. Access to the radio links is controlled by Layer 2 access methods used by all radios in the Voice Net Segment, and by human protocols. Communications are half-duplex either by design or by enforcement of human protocol. Other Voice Nets support point-to-point communications initiated by one radio connecting to one or a few designated radios using full duplex communications. This version of the UCR is limited to describing requirements for the support of PTT-based voice networks.

Voice Net technology is not standardized. It is not the purpose of this UCR to create such standards. The requirements in this UCR are directed toward the communications between a VNAR and a UC-compliant voice EI. The basic requirement is that a VNAR provide a standardized EI interface so that other EIs can connect to the VNAR and become parties to the Voice Net. Figure A.6-1, TRN Connectivity, shows the VNAR connected directly to an Assured Services LAN (ASLAN). However, the VNAR also could connect via the RAN to a device that connects to an ASLAN.

A.6.5 Functional Description

This appendix defines the requirements for VNAR UCR compliance and requirements changes to enable Strategic and Deployed SCs to support traffic flow between UCR-compliant EIs and VNARs.

Figure A.6-2, Functional Connectivity, provides an overview of the major functions necessary to provide connectivity between VoIP EIs and a TRN. This figure does not include the specifics of how the functions connect to each other.

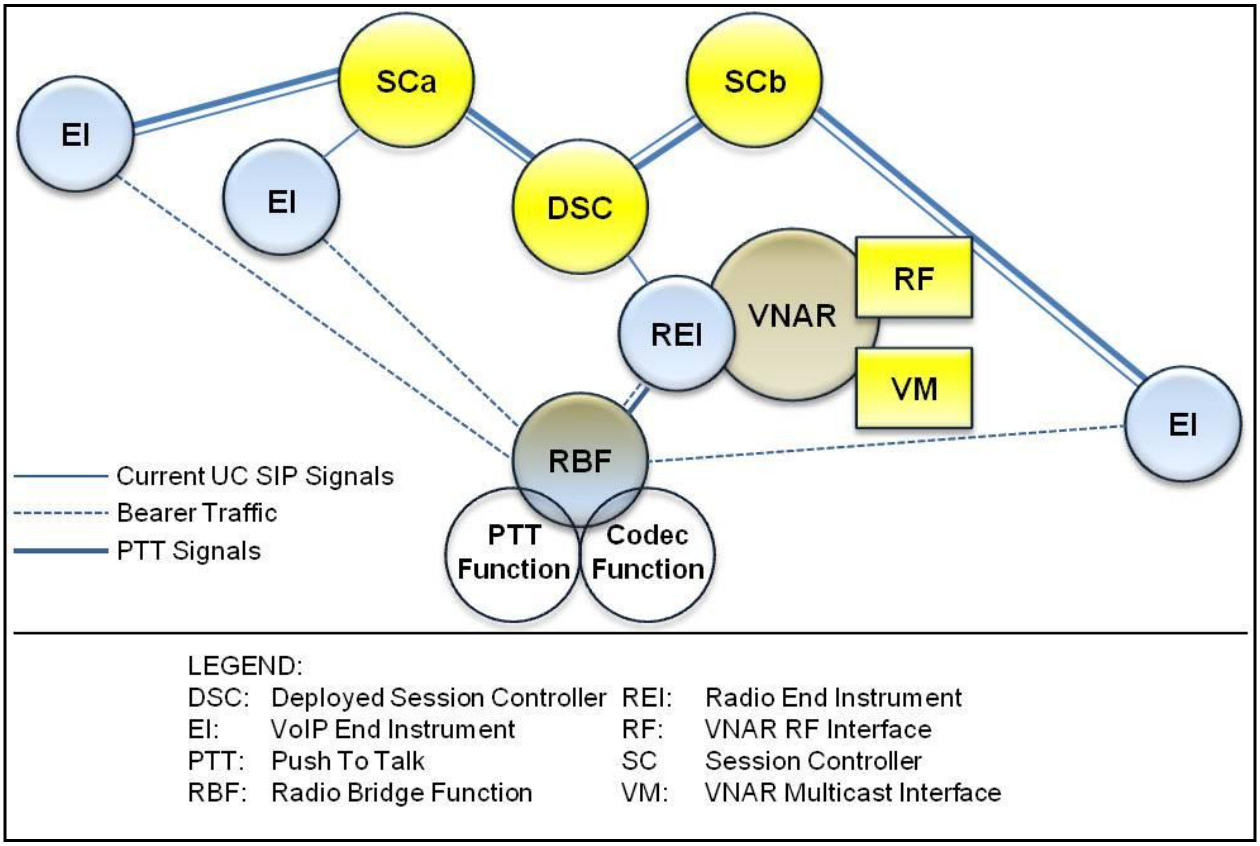


Figure A.6-2. Functional Connectivity

The flow, shown in Figure A.6-2, Functional Connectivity, assumes that PTT signaling will be sent out-of-band, via the SCs using the concepts defined in RFC 4730. Alternatively, the PTT signaling could be sent in-band, using the concepts defined in RFC 4733. In such case, the tone signals will be sent along the bearer path between the EI (or a proxy for the EI) and the RBF.

The EIs and SCs connect over a UCR-compliant IP network, as indicated in Figure A.6-1, TRN Connectivity. The EIs and SCs perform the functions defined for these elements as described within this UCR in Section 2 and in UC Framework 2013, Section 2. In addition, the EIs and SCs shall be enhanced to support a new set of UC SIP signals that support PTT requirements for Tactical radios. The SCa and SCb in Figure A.6-2 support fixed EIs. The deployed SC (DSC) in the figure supports deployed TRNs and cellular networks as described in Section A.5, DCVX System, and for DSC [to be determined (TBD)].

The RBF provides connectivity between the EIs and the Radio End Instrument (REI). It is an enhanced version of a conventional conference bridge described in Section 3.4, UC Audio and Video Conference System.

The REI provides connectivity between the UCR-compliant domain and the Voice Net. The connection could require multiple router and switch hops. The RBF and REI also connect to the DISN IP voice network. The connection could be a LAN, WAN, or a back plane (if the functions are collocated in the same physical device).

The RBF could be a stand-alone appliance, or incorporated in a DSC or incorporated in a VNAR. The PTT function, associated with the RBF, emulates the PTT function of the TRN to enable a conference participant to access the Voice Net. The codec function associated with the RBF performs transcoding as necessary, to enable media transfer between the REI and the EIs. The REI is always located within a VNAR. The VNAR is connected to an RF element, and possibly a RAN.

The EIs and REIs join a conference, which is dedicated to the Voice Net. The RBF provides half-duplex, PTT access for the EIs to the Voice Net and enables a conference participant to speak to the Voice Net. The UC SIP is enhanced to provide a new set of signaling functions that support PTT access as described in Table A.6-1, Control Information: VNAR to VoIP EI, and Table A.6-2, Control Information: VoIP EI to VNAR.

Voice Net bearer traffic, shown as dotted lines in Figure A.6-2, Functional Connectivity, flows between the REI and the RBF where it is replicated for transmission to each EI that has been admitted to the conference. Bearer traffic flows from each EI to the RBF. In general, this traffic is blocked at the RBF. The only exception is bearer traffic from an EI that has been granted access via the PTT function.

Current UC SIP signaling traffic is shown as solid lines in Figure A.6-2, Functional Connectivity. The PTT signaling traffic is shown as double lines in Figure A.6-2, Functional Connectivity. Both types of signal traffic flow between an EI and its Master SC (MSC), from SC to DSC to the RBF and the REI and in the reverse direction. The PTT signaling traffic also flows in both directions between the RBF and the REI.

A.6.5.1 Radio Bridge Function

TAC-000700 [**Required: RBF**] The system shall support three types of participants in a "Meet-Me Bridge" Voice Net conference (see Section 3.4, UC Audio and Video Conference System):

a. Conference Participant. An individual who joins the Voice Net using VoIP EIs.

b. VNAR Participant. One or more VNARs connected to the Voice Net.

c. Conference Manager. A conference participant who has the authority to manage certain features of the conference.

Each participant joins the conference by calling in to a unique telephone number or Uniform Resource Identifier (URI) that is assigned to the conference. The RBF will be assigned a unique IP address that corresponds to its telephone number and URI.

TAC-000710 [**Required: RBF**] The system shall lead the caller through an authentication process using voice messages. If the process determines that the caller is authorized to join the conference, then the system shall send a voice message informing the caller that he or she is now a conference participant and indicate the status of the conference. Status shall include the following:

a. Conference is not yet available.

b. Conference has been terminated.

c. Conference is in process including the number of participants, and an indication if the VNAR is a participant.

TAC-000720 [**Required: RBF**] If the authentication process determines that the caller is unauthorized, then the system shall send a voice message informing the caller that he or she is ineligible for the conference. If the caller does not hang up in a parameter-determined period, then the system will terminate the call, whereas the parameter termination period shall be determined using a configurable time-out parameter with a time-out range of 0–60 seconds; default shall be set to 5 seconds.

TAC-000730 [**Required: RBF**] The authentication process shall include a participant code, which identifies the type of participant, and a log-in process suitable to the security level of the conference.

TAC-000740 [**Required: RBF**] The system shall perform the following functions in addition to those described in Section 3.4, UC Audio and Video Conference System:

a. In the default mode, a conference participant is placed in a listen-only mode, where the participant can only hear audio transmitted from the VNAR.

b. The system performs whatever codec transformations are necessary to ensure compatible communications between the VNAR and the EIs (see Section A.6.5.4, Codec Translation Functional).

c. The system supports the PTT function defined in Section A.6.5.3, Push-To-Talk Functional. The PTT function will ensure that only one conference participant can speak to the Voice Net at a time, and only when there is no other party speaking on the Voice Net (see Section A.6.5.3).

d. The conference manager has the ability to block or preempt any participant from access to the Voice Net.

e. The conference manager has the ability to bridge the conference participants to each other, so they can speak with each other. The traffic resulting from this bridging will not be transmitted to the Voice Net.

f. The conference manager has the ability to speak to any or all conference participants.

g. The conference manager has the ability to terminate the speaker-listener status of any conference or VNAR participant.

TAC-000750 [**Required: RBF**] The system shall operate as an UC SIP EI for authenticating, registering, and interacting with the SC to originate or terminate voice sessions.

NOTE: The RBF exchanges UC SIP signaling packets with its Master DSC. The DSC exchanges UC SIP messages with other SCs to complete and tear down calls. The

Master DSC could be collocated with or remote from the RBF. The Master DSC could be assigned uniquely to the RBF, or could support multiple RBFs.

TAC-000760 [**Required: RBF**] The system shall provide a method for multiple VoIP EI users to concurrently connect to the Voice Net, up to a configurable limit.

TAC-000770 [**Required: RBF**] The system shall allow automatic termination of the session, based on configurable events, including inactivity on an UC SIP session for a specified session time limit.

TAC-000780 [**Required: RBF**] The system shall support MLPP requirements if more calls arrive than can be supported (see Section 2.25.2, Multilevel Precedence and Preemption). The system shall be able to preempt a call from a lower precedence conference participant, if necessary, to provide resources to accept a call from a higher precedence conference participant if that call would otherwise be blocked. The system shall not preempt a call from a VNAR participant unless directed to do so by the conference manager.

TAC-000790 [**Required: RBF**] The system shall support a configurable number of simultaneous conference participants per Voice Net.

TAC-000800 [**Required: RBF**] The system shall support a configurable number of simultaneous VNAR participants per Voice Net.

TAC-000810 [**Required: RBF**] The conference manager shall select one VNAR to act as master.

TAC-000820 [**Required: RBF**] When the system is not acting as a master, the system shall act as a backup, which is available to replace the master system if the master fails. It is highly desirable that the system be implemented to support an automatic failover to a backup VNAR if a master VNAR or its connection fails.

NOTE: The number of VNAR and conference participants is unspecified. These numbers are left to best design and engineering practices as determined by the supplier of the RBF function to meet the performance and reliability goals required for a particular deployment.

A.6.5.2 Radio End Instrument

TAC-000830 [**Required: REI**] The system which resides within the VNAR, shall act as a conventional UCR-compliant EI in performing the following features defined in the following sections:

a. Point-to-Point Call (UC SIP 2013, Section 9.2, Point-to-Point Call, and UC SIP 2013, Section 16.2, Point-to-Point Call).

b. Tracing of terminating call (see Section 2.2.11.2, Tracing of Terminating Calls).

c. Outgoing call tracing (see Section 2.2.11.3, Outgoing Call Tracing).

d. Tracing of a call in progress (see Section 2.2.11.4, Tracing of a Call in Progress).

TAC-000840 [**Required: REI**] The system shall transform bearer traffic packets received from the RBF to the format associated with its Voice Net and transmit the traffic to the Voice Net.

TAC-000850 [**Required: REI**] The system shall transform bearer traffic received from the Voice Net to bearer traffic packets for transmission to the RBF.

TAC-000860 [**Required: REI**] The system shall transform UCR standard PTT signals defined in Section A.6.5.3, Push-To-Talk Functional, to the PTT signals required by the VNAR to support the Voice Net.

TAC-000870 [**Required: REI**] The system shall transform PTT signals received from the Voice Net to UCR standard PTT signals for transmission to the RBF.

TAC-000880 [**Required: REI**] The system shall operate as an UC SIP EI for the purpose of authenticating, registering, and interacting with its Master DSC to originate or terminate voice sessions.

NOTE: The DSC could be collocated with or remotely from the REI. The DSC could be assigned uniquely to the REI, or it could support multiple Voice Nets. The DSC provides an MLPP function to give priority to higher precedence callers, if there is insufficient capacity to support a new dial-in call.

TAC-000890 [**Required: REI**] The system shall support incoming session setup requests from IP EIs according to the UC SIP specification [Reference: "Department of Defense Assured Service Session Initiation Protocol (UC SIP) Generic System Requirement (GSR)"].

TAC-000900 [**Conditional: REI**] The system shall support a call to a VoIP EI directly without going through an external RFB. If this option is invoked, then the REI shall support the following:

a. The PTT function described in Section A.6.5.3, Push-To-Talk Functional.

b. The codec translation function described in Section A.6.5.4, Codec Translation Functional.

c. The MLPP functions described in Section 2.26.2, Multilevel Precedence and Preemption.

d. Three-way calling as described in Section 2.2.6, Three-Way Calling.

A.6.5.3 Push-To-Talk Functional

The VNAR, RBF, SCs, and REIs cooperate to provide a capability that will enable a VoIP end user to initiate and terminate the equivalent of a PTT session.

TAC-000910 [**Required: RBF**] The system shall provide a fail-safe mechanism to prevent a VoIP EI from streaming continuous voice traffic to a PTT-based Voice Net.

TAC-000920 [**Required: RBF**] The system fail-safe mechanism shall ensure that, no matter the status of the VoIP end user or the VoIP EI, transmissions from the VoIP EI will terminate within a configurable, parameter-driven amount of time.

TAC-000930 [**Required: RBF**] The system fail-safe mechanism shall only reinstate transmissions based on completion of a specific, positive action by the VoIP end user.

TAC-000940 [**Required: VNAR, RBF, REI – Conditional: SC or DSC**] The cooperating elements in a PTT session shall support Dual Tone Multifrequency (DTMF) tones after a call has progressed to the media session mode.

TAC-000950 [**Required: RBF**] If PTT is configured, then the system shall prevent bearer traffic generated from a VoIP EI, from accessing the Voice Net until the VoIP end user initiates a PTT session by entering a unique configurable tone sequence called the "talk tone." This action mimics depression of the PTT button on a radio, thereby initiating an emulated PTT session. The VoIP end user enters a different, configurable tone sequence ("end tone") to end the PTT session, emulating the release of the PTT button.

TAC-000960 [**Required: RBF**] Upon receipt of the Talk Tone, the system in cooperation with the REI, shall determine whether the Voice Net is busy or available. The Voice Net is busy if any other party has been granted authorization to speak on the Voice Net. This can happen in one of two ways: 1) another conference participant has access to the Voice Net, or 2) a radio user has access to the Voice Net.

The following examples present traffic flow for the two cases where the Voice Net is busy, and the case where the Voice Net is available. These examples are representative, but not exclusive. Traffic flows could vary based on the type of technology used in the Voice Net. The examples assume out-of-band signaling and the use of VNARs that can provide tone responses indicating a Voice Net available condition. The flow of traffic for the first busy case is as follows:

1. The participant keys in the talk tone sequence.
2. The talk tone is sent from the originating EI to its MSC.
3. The SC converts the EI-generated talk tone to the UCR standard talk tone packet.
4. The SC sends the talk tone packet to the DSC to which the RBF is registered.
5. The RBF's Master DSC sends the talk tone packet to the RBF.
6. The RBF determines that another conference participant has access to the Voice Net and cannot be preempted.
7. The RBF generates a standard "busy tone" packet for transmission to its Master DSC.
8. The DSC sends the busy tone packet to the SC to which the requesting EI is registered.

9. The EI's MSC converts the tone to a form that is supported by the requesting EI.
10. The requesting EI creates an audio signal indicating that the Voice Net is busy.

The flow of traffic for the second busy case is as follows:

1. The participant keys in the talk tone sequence.
2. The talk tone is sent from the originating EI to its MSC.
3. The SC converts the EI-generated talk tone to the UCR standard talk tone packet.
4. The SC sends the talk tone packet to the DSC to which the RBF is registered.
5. The Master RBF sends the talk tone packet to the RBF:
   a. The RBF determines that no other conference participant has access to the Voice Net.
   b. The RBF sends the talk tone packet to the REI.
   c. The REI translates the information in the packet to the form required by the VNAR to access the Voice Net and the Voice Net sends a signal back to the VNAR indicating that the Voice Net is busy.
   d. The REI, within the VNAR, generates a standard busy tone packet and sends it to the RBF.
   e. The RBF sets a flag indicating that the Voice Net is busy and sends the standard busy tone packet to its Master DSC.
   f. The DSC sends the busy tone packet to the EI's MSC.
   g. The SC converts the tone to a form that is supported by the requesting EI, and sends the tone to the EI.
   h. The requesting EI creates an audio signal indicating that the Voice Net is busy.

   NOTE: There is a timing issue in the busy cases. In some situations, the delay between the time of the initial VoIP EI PTT request and the time the request arrives at the VNAR could be in the 1–2 second range. This could lead to a situation where speakers on the TRN could block out VoIP speakers. To mitigate this situation, as an optional feature, the REI could store a blocked request from a VoIP EI, wait until the Voice Net is available, and then initiate the request to the Voice Net. If this occurs, then the REI would send a busy tone immediately followed by an available tone to the EI, followed by periodic busy tones. When the Voice Net becomes available to the EI, the REI will send a burst of two available tones to the EI.

If the Voice Net is not busy, then it is considered available.

The signaling sequence for an available Voice Net, assuming out-of-band signaling is as follows:

1. The participant keys in the talk tone sequence.

2. The talk tone is sent from the originating EI to its MSC.
3. The MSC converts the talk tone to the UCR-standard talk tone packet.
4. The MSC sends the talk tone packet to the DSC to which the RBF is registered.
5. The RBF's Master DSC sends the talk tone packet to the RBF.
6. The RBF determines that no other conference participant has access to the Voice Net.
7. The RBF sends the talk tone packet to the REI.
8. The REI translates the information in the packet to the form required by the VNAR to access the Voice Net.
9. The REI translates the information in the packet to the form required by the VNAR to access the Voice Net and the Voice Net sends a signal back to the VNAR indicating that the Voice Net is available.
10. The REI, within the VNAR, formats a standard "Voice Net Available" packet and sends it to the RBF.
11. The RBF performs the following functions:
    a. Sets a flag indicating that the Voice Net is busy.
    b. Sends the Voice Net Available packet to its Master DSC.
    c. Starts a PTT timer based on a configurable parameter.
    d. Enables voice traffic from the originating EI to flow to the REI.
    e. Blocks traffic from the Voice Net to the originating EI.
12. The DSC sends the Voice Net Available packet to the requesting EI's MSC.
13. That SC converts the tone to a form that is supported by the requesting EI and transmits the tone to the EI. The SC also signals the EI to return to the media transmission mode.
14. The requesting EI creates an audio signal indicating that the Voice Net is available and reverts to the media mode of operation.
15. The conference participant can now speak. Bearer traffic will be sent from the EI to the RBF. The RBF will send the bearer packets to the conference participants and the REI. The REI will translate the bearer traffic so that the VNAR can transmit the bearer traffic to the Voice Net.

TAC-000970 [**Required: VNAR, RBF, REI – Conditional: SC or DSC**] The PTT session shall terminate upon any of the following activities:

a. The VoIP end user keys in the end tone. This tone is sent to the MSC and from there to the SC serving the RBF and from there to the RBF.
b. A voice activity detection (VAD) device in the RBF determines that there has been no voice activity for a configurable time.

c. The PTT timer expires.

TAC-000980 [**Required: RBF**] Upon termination of the PTT session, the system shall execute the following actions:

a. Transmit a "PTT Terminated" tone packet to the VoIP EI (via the appropriate MSCs) for a configurable amount of time.
b. Reset the PTT timer.
c. Reset the Voice Net busy flag.
d. Re-enable the transmission of bearer traffic from the Voice Net to the VoIP EI.
e. Send a PTT Terminated packet to the REI.
f. The REI will convert the information in the PTT Terminated packet to the form required by the VNAR to terminate the PTT session in the Voice Net.
g. The Voice Net will become available to other parties who wish to speak.

TAC-000990 [**Required: RBF**] At some configurable time before a PTT time-out, the system shall issue a "Warning" packet to inform the speaker the session is about to terminate. The Warning packet will be transmitted via the signaling path, from the system's DSC to the conference participant's SC to the EI.

TAC-001000 [**Required: RBF**] The system shall ignore a talk tone generated by a VoIP EI that is in a PTT session.

TAC-001010 [**Required: RBF**] The system shall ignore an end tone generated by a VoIP EI that is not in a PTT session.

TAC-001020 [**Objective: VoIP EI**] It is desirable if the VoIP EI could be modified to include a special control key that must be depressed to maintain the emulated PTT session. This would emulate the PTT action at a radio more accurately, and potentially reduce dead time associated with the use of a timer.

TAC-001030 [**Required: RBF**] The system shall provide the following configurable mechanisms to mitigate situations where the VoIP EIs might not support tone patterns to define the beginning and end of a PTT session:

a. The conference manager shall have the ability to place the VoIP EI in listen-only mode.
b. The system shall transmit a "warning tone" to the VoIP EI if there is voice traffic generated from a radio on the Voice Net.
c. The system shall invoke a configurable off-on feature, which will limit the time duration of transmissions from the VoIP EI. The system shall not forward traffic from such devices for more than the configurable amount of time. If the system terminates a PTT session based on this parameter, then it shall not forward traffic from the VoIP EI until a configurable amount of time has passed since the end of the last transmission period.

TAC-001040 [**Required: RBF**] Table A.6-1, Control Information: VNAR to VoIP EI, defines the standard tones that shall be used to convey status information to the VoIP EIs.

Table A.6-1. Control Information: VNAR to VoIP EI

CONTROL SIGNAL ID	SIGNAL NAME	SIGNAL CONFIGURATION (TONES)	DESCRIPTION	ACTION
1	Voice Net Available	TBD	The Voice Net is operational and is not busy.	Caller can start to talk. Typically sent in response to a talk tone request sent by the caller (See Table A.6-2).
2	Voice Net Operational But Not Available	TBD	The Voice Net is operational but cannot be accessed because of a lack of resources to support call; also used to indicate that the call has been preempted.	Caller should hang up and try again later. Typically sent in response to a talk tone.
3	Voice Net Busy –"Busy Tone"	TBD	The Voice Net is operational and reachable, but is busy.	Caller should try again later. Typically sent in response to a talk tone.
4	PTT Terminated	TBD	The VNAR has terminated the PTT session based on a request from the VoIP caller or a time-out.	For information purposes; caller should stop talking.
5	Voice Net Secure	TBD	Transmissions from the VNAR to the Voice Net are sent in encrypted or scrambled mode.	For information purposes. Typically sent in response to a talk tone.
6	Voice Net Plain Text	TBD	Transmissions from the VNAR to Voice Net are sent in plain text mode.	Caller should not talk if he or she were expecting that transmissions would be secure. Typically sent in response to a talk tone.
7	Stop Transmitting	TBD	Stop talking: either the speaker is talking for too long, or there is a higher priority speaker who needs access to the Voice Net.	The VNAR will block voice from reaching Voice Net. Caller should stop talking.
8	Warning	TBD	The VNAR will terminate voice traffic from VoIP EI within a configurable period.	Caller should be aware that he or she will soon get a stop transmitting signal.
LEGEND ID: Identification PTT: Push-to-Talk			TBD: To Be Determined VNAR: Voice Net Access Radio	

NOTE: The typical Voice Net will not generate all the control signals shown in Table A.6-1, Control Information: VNAR to VoIP EI. It is up to the designer of the VNAR to determine which control signals must be implemented. However, if a signal shown in Table A.6-1 is used, the tones transmitted shall be the ones defined in Table A.6-1. It is possible that future generations of TRNs will create additional signals. If required, this UCR will be modified to accommodate the new signals.

Table A.6-2, Control Information: VoIP EI to VNAR, defines the signals and standard tones that shall be used at the VoIP EI to indicate the start and termination of a PTT session.

Table A.6-1, Control Information: VNAR to VoIP EI, and Table A.6-2, Control Information: VoIP EI to VNAR, do not include UC SIP signaling, which is discussed in Section 2, Session Control Products.

Table A.6-2. Control Information: VoIP EI to VNAR

CONTROL SIGNAL ID	SIGNAL NAME	SIGNAL CONFIGURATION (TONES)	DESCRIPTION	ACTION
1	PTT Request – "Talk Tone"	TBD	A tone that indicates the start of a PTT session.	The VNAR will format and initiate PTT request to VNAR, if the Voice Net has PTT capability.
2	PTT Terminate – "End Tone"	TBD	A tone that indicates the termination of a PTT session.	The VNAR will terminate PTT request to Voice Net.
LEGEND ID: Identification PTT: Push-to-Talk			TBD: To Be Determined VNAR: Voice Net Access Radio	

A.6.5.4 Codec Translation Functional

TAC-001050 [**Required: REI**] The system shall support at least one of the following Real-Time Services (RTS) VoIP codecs:

a. G.711a and µ-law (64 kbps).

b. G.723.1 (5.3 and 6.3 kbps).

c. G.729d (6.4 kbps).

d. G.729e (12.4 kbps).

TAC-001060 [**Required: REI**] The system shall support the Enhanced Mixed Excitation Linear Production (MELPe) codec at 2.4 Kbps and lower rates.

NOTE: When a call is set up between an RTS VoIP EI and the RBF, the codec function will negotiate the codec using the SDP. When a call is set up between an REI and the RBF, the codec function will negotiate the codec using the SDP. The codec function will attempt to minimize the bandwidth required between the RBF and the REI.

TAC-001070 [**Required: RBF**] The codec function is always associated with the RBF and will reside in the same device as the RBF.

A.6.5.5 End Instrument Functional Changes

TAC-001080 [**Optional: AEI, PEI**] The system shall support the PTT signaling functions described in Section A.6.5.3, Push-To-Talk Functional.

TAC-001090 [**Objective**] It is highly desirable that the EIs be configured to directly process the standard PTT signaling packets used to create the audio tones and convey user keystrokes during a PTT session.

TAC-001100 [**Optional**] It is permissible to have the EI's MSC create the standard signaling packets and use a nonstandard approach to conveying that signaling information between the EI and its Master SC.

TAC-001110 [**Objective**] It is highly desirable that EI PTT signaling support be implemented by software and configuration downloads, rather than hardware changes in the EIs.

A.6.5.6 SC Functional Changes

TAC-001120 [**Optional: SC**] The system shall support the PTT signaling function additions to UC SIP described in Section A.6.5.3, Push-To-Talk Functional. If the EI does not directly create and process standard PTT signaling packets, then it is highly desirable that the SCs be able to download software changes to the EIs as described in Section A.6.5.5, End Instrument Functional Changes.

A.6.6 Network Management

General NM requirements are specified in Section 2.17, Management of Network Appliances.

TAC-001130 [**Required: RBF**] The system shall identify the unique name of the TRN supported by the conference.

A.6.7 Tactical LAN

The assured services objectives are difficult to achieve in the Tactical-edge networks (TNs), due to dynamically changing connectivity, limited bandwidth, unstable environment, and limited equipment. The TNs are often non-ASLAN compliant.

The UCR does not permit non-ASLAN-compliant devices to support special C2 and C2 users. This architecture has to be modified to allow an EI located at a non-ASLAN Tactical location to support special C2 and C2 users, provided a Type I encryption mechanism is applied to the call signaling messages and bearer traffic.

A.6.7.1 Physical Media

TAC-001140 [**Required: VNAR**] The system must support at least one of the following Ethernet types:

a. 10 Base-T.
b. 100 Base-T.
c. 1000 Base-T.

A.6.7.2 Dial Plan and Routing

(Reference Section 2.16, Worldwide Numbering and Dialing Plan.)

TAC-001150 [**Required**] Each Voice Net shall be assigned a routable user identity, which can be one of the following: DSN number, Tel-URI, SIP-URI, and FQDN.

A.6.7.3 DSCP

TAC-001160 [**Required: RBF, REI, VNAR**] The system shall provide a configurable mechanism to mark DSCPs in the header of IP packets. The default marking shall be as defined in Section 6, Network Infrastructure End-to-End Performance.

A.6.7.4 Traffic Engineering

The REI supports only voice and control traffic, and can apportion that traffic in any manner as determined by traffic engineering. The number of subscribers that need to be supported will be determined by each Program Office (PO).

The VNAR should operate within the overall network voice E2E delay and jitter requirements as specified in the UCR and UC Framework, Section 6, for end-to-end service level requirements, and voice requirements, specifically Section 7.3.1, Voice Services. The RECOMMENDED upper limit on the average post-selection delay for various E2E scenarios is defined in Table A.6-3, Upper Limit on Average Post-Selection Delay.

NOTE: The UCR and REI time delays stated in Table A.6-3 relate only to the time it takes to set up a call to the RBF. Typically, this is done once per conference per VoIP EI and REI. The PTT signaling requests will occur many times during the conference. The time to implement a PTT request from a VoIP EI and return an available signal to the EI involves round-trip delay between the EI and the REI. Times will vary from subseconds if there are no satellite links involved, to as many as 2–3 seconds, if satellite links are involved. The maximum delay to release an EI-initiated access to the TRN is determined by one-way delay and is proportionately less.

Table A.6-3. Upper Limit on Average Post-Selection Delay

TYPE OF DELAY	UCR	REI
Local intratheater DSN call signaling during normal network traffic load	1 second	TBD
Local intrabase DSN call signaling during normal network traffic load	1.5 seconds	TBD
Worldwide DSN call signaling during normal network traffic load	6.0 seconds	TBD
Global DSN call signaling during normal network traffic load	8.0 seconds	TBD
LEGEND DSN: Defense Switched Network TBD: To Be Determined REI: Radio End Instrument UCR: Unified Capabilities Requirements		

A.7 DEPLOYED (TACTICAL) MASTER SC AND SUBTENDED SC REQUIREMENTS AND DYNAMIC ASAC (DASAC) REQUIREMENTS IN SUPPORT OF BANDWIDTH CONSTRAINED LINKS

Since these requirements are applicable to the Fixed (Strategic Enterprise), as well as to the Deployed (Tactical) environment, these requirements are defined in Section 2.24, Master SC and Subtended SC Requirements and Dynamic ASAC (DASAC) Requirements in Support of Bandwidth Constrained Links. Many of these requirements, which are mandatory for the Deployed environment, are conditional for the Fixed environment.

A.8 LAN/WAN OPTIMIZER

A.8.1 Introduction

This document establishes a new product category within the UCR. This product category defines the functions and requirements specific to a Wide Area Network (WAN) Optimization Controller (WOC).

A.8.1.1 Purpose

WOCs fall within the general class of NEs. WOCs perform Traffic Conditioning Services, Bandwidth Management Services, and Per-Hop Behavior (PHB) Management Services in order to achieve an improved level of performance in the transport efficiency of data. To achieve the desired level of traffic conditioning, various techniques are leveraged to manipulate processes at various layers within the Open System Interconnect (OSI) Model.

Optimization may be sought for a number of reasons; high latency links, bandwidth constrained links, or to overcome the effects of excessively “chatty” applications. LAN/WAN Optimizers can now be connected to the DISN in order to optimize Internet Protocol (IP) network capabilities. To take advantage of these advancements, this appendix defines the LAN\WAN Optimizer requirements that must be met in order to be placed on the UC APL. The goal of optimization is to improve network efficiency and performance due to faster IP transport and improved bandwidth utilization.

A.8.1.2 Applications and Configurations

WOCs address four distinct data transport configurations based on the physical characteristics of the WAN infrastructure, its primary function, or its method of implementation. The four configurations and transport characteristics are identified as follows:

- Transport:
  - Satellite Network (SN). High latency because of a long physical propagation time; bandwidth is limited.
  - Terrestrial Network (TN). Users often experience slow network responsiveness.
  - Configurations.
  - Disaster Recovery (DR). Requires speedy transport of high volume traffic.
  - Software Clients. Software- based optimization capability. Software Clients can support individual remote or mobile users, or multiple users depending on application.

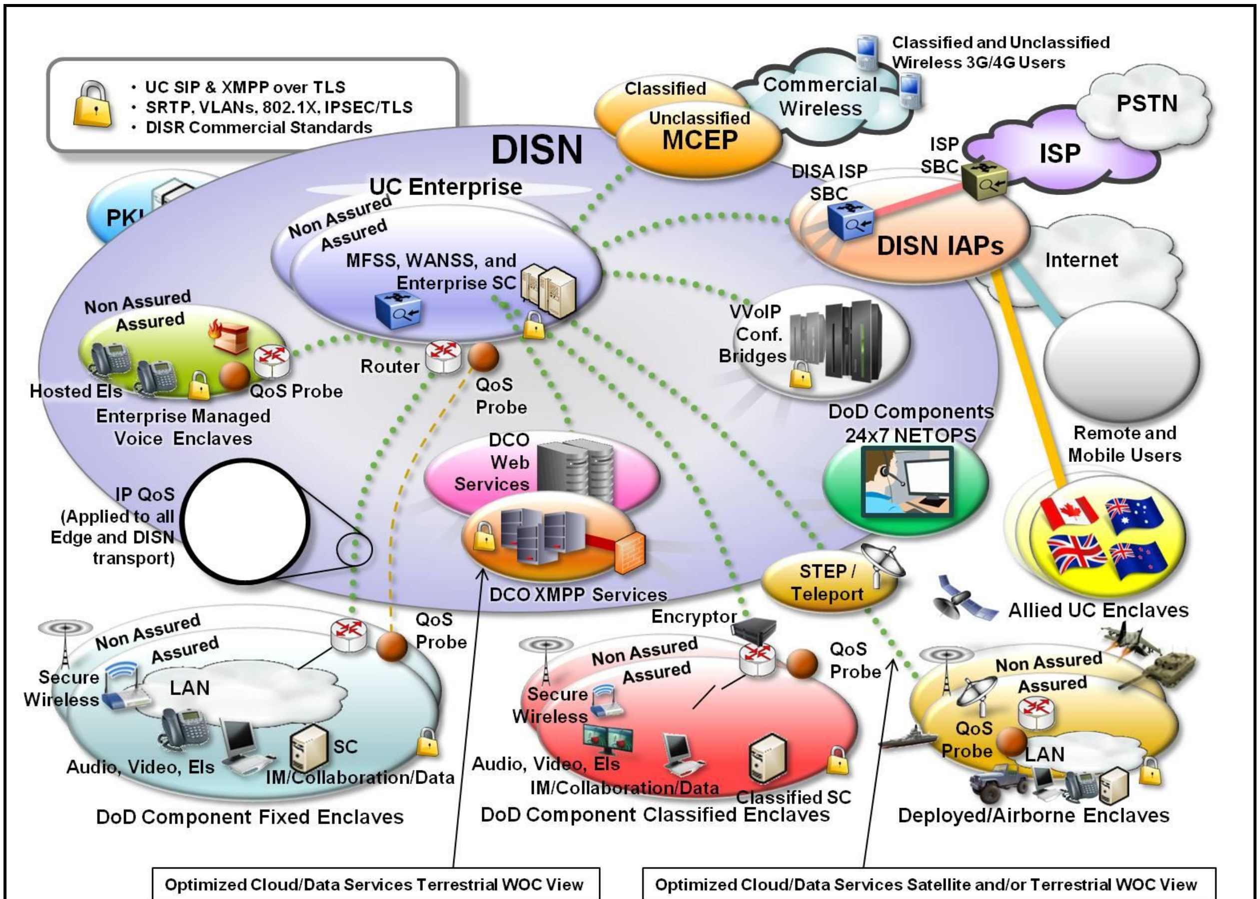
A.8.1.3 WOC Functional Description

WOCs perform specific traffic conditioning processes to improve delivery time and bandwidth utilization across LAN/WAN infrastructures. Typically, these processes are combinations of techniques meant to improve the performance at several layers in the OSI model. These improvements are usually achieved by modifications in the TCP/IP model and or the OSI model. There are two distinct modifications in use:

Transport Protocol-optimizations operate primarily at layer 4 and tend to focus on streamlining TCP and other protocol chattiness to overcome latency issues. Optimizations are achieved via Selective Acknowledgment (SACK), Space Communications Protocol Standards (SCPS), Window Sizing, Congestion Avoidance Modification, etc.

Application layer optimizations usually operate at several OSI layers simultaneously, typically Layers 5–7, to achieve improved performance of application layer processes and user activities.

WOC optimizers are normally deployed in pairs. In tandem, they perform all of the functions required to optimize the prevailing circuit conditions for the IP traffic type that the WOC is transporting; this relationship is depicted in the DISN architecture. Figure A.8-1, UC Operational Framework.



LEGEND:

3G:	3rd Generation	IA:	Information Assurance	MFSS:	Multi-Function Softswitch	TLS:	Transport Layer Security
4G:	4th Generation	IAP:	Internet Access Portal	NetOps:	Network Operations	UC:	Unified Capabilities
Conf:	Conference	IM:	Instant Messaging	PKI:	Public Key Infrastructure	UC SIP:	Unified Capabilities Session Initiation Protocol
DCO:	Defense Connect Online	IP:	Internet Protocol	PSTN:	Public Switch Telephone Network	VLAN:	Virtual Local Area Network
DISN:	Defense Information Systems Network	IPSec:	Internet Protocol Security	SBC:	Session Border Controller	WAN:	Wide Area Network
DISR:	DOD Information Technology Standards Registry	ISP:	Internet Service Provider	SRTP:	Secure Real-Time Transport Protocol	WOC:	WAN Optimization Controller
DOD:	Department of Defense	LAN:	Local Area Network	SS:	Softswitch	XMPP:	Extensible Messaging and Presence Protocol
EI:	End Instrument	MCEP:	Multi-Carrier Entry Point	STEP:	Standardized Tactical Entry Point	WoIP:	Wireless over IP

Figure A.8-1. UC Operational Framework

A.8.2 WOC Interfaces

WOC interfaces can be divided into three categories: Ingress (or LAN-side), Egress (or WAN-side), and management.

A.8.2.1 Ingress Interface(s)

This network interface connects to a non-Assured Services LAN (non-ASLAN) or an ASLAN. The WOC must minimally provide the required ingress interfaces listed. The WOC may conditionally provide other interfaces.

TAC-001170 [**Required**] The WOC shall minimally provide one of the following ingress interface rates that are capable of being optimized. Interface should be able to auto negotiate down to the lowest stated rate if necessary. (Other rates and Institute of Electrical and Electronics Engineers (IEEE) standards may be provided as conditional interfaces):

a. 100 Megabits per second (Mbps) IAW IEEE 802.3i.

b. 10 Mbps IAW IEEE 802.3j.

c. 100 Mbps IAW IEEE 802.3u.

d. 1000 Mbps IAW IEEE 802.3z.

e. 1000 Mbps IAW IEEE 802.3ab.

TAC-001180 [**Optional**] The WOC shall provide a minimum of one (1) ingress bypass port from the following interfaces (other rates and IEEE standards may be provided as conditional interfaces). By-pass requirements are specified in Section A.8.3.4, Transparency:

a. 10 Mbps IAW IEEE 802.3i.

b. 10 Mbps IAW IEEE 802.3j.

c. 100 Mbps IAW IEEE 802.3u.

d. 1000 Mbps IAW IEEE 802.3z.

e. 1000 Mbps IAW IEEE 802.3ab.

A.8.2.2 Egress Interface(s)

This network interface connects to the WAN. The WOC must minimally provide the required egress interfaces listed. The WOC may conditionally provide other interfaces.

TAC-001190 [**Required**] The WOC shall minimally provide one of the following egress interface rates that have been optimized (other rates and IEEE standards may be provided as conditional interfaces). Optimized requirements are specified in Section A.8.3.2, Optimization:

a. 10 Mbps IAW IEEE 802.3i.

b. 10 Mbps IAW IEEE 802.3j.

c. 100 Mbps IAW IEEE 802.3u.

d. 1000 Mbps IAW IEEE 802.3z.

e. 1000 Mbps IAW IEEE 802.3ab.

TAC-001200 [**Conditional**] The WOC shall provide a minimum of one (1) egress bypass port from the following interfaces. By-pass requirements are specified in Section A.8.3.4, Transparency:

a. 10 Mbps IAW IEEE 802.3i.

b. 10 Mbps IAW IEEE 802.3j.

c. 100 Mbps IAW IEEE 802.3u.

d. 1000 Mbps IAW IEEE 802.3z.

e. 1000 Mbps IAW IEEE 802.3ab.

A.8.2.3 Management Interface(s)

The Network management interface provides a dedicated management interface. The WOC shall minimally provide the following management interface.

TAC-001210 [**Required**] The WOC shall minimally provide one of the following management interface rates that have been optimized (other rates and IEEE standards may be provided as conditional interfaces):

a. 10 Mbps IAW IEEE 802.3i.

b. 10 Mbps IAW IEEE 802.3j.

c. 100 Mbps IAW IEEE 802.3u.

d. 1000 Mbps IAW IEEE 802.3z.

e. 1000 Mbps IAW IEEE 802.3ab.

A.8.3 Functional

The requirements in this appendix specify functional behaviors applicable to the WOC process. In addition to the requirements listed below, WOC devices must meet the requirements identified in UCR 2013, Section 11.2, DSN Fixed Network Element (F-NE) Generic, and 11.3, Deployed Network Element (D-NE), as specified in the following text. The requirements in this appendix apply to all WOC devices to include hardware based solutions as well as virtualized solutions via software.

A.8.3.1 WOC Network Element (NE)

TAC-001220 [**Required**] The WOC shall meet the NE requirements specified in Section 11.2, DSN F-NE Generic, for all WOCs deployed in strategic terrestrial configurations; or

TAC-001230 [**Required**] The WOC shall meet the NE requirements specified in Section 11.3, D-NE, for tactical configurations.

A.8.3.2 Optimization

TAC-001240 [**Required**] When the WOC is deployed in pairs across satellite communications links it shall allow for utilization of SCPS protocol.

TAC-001250 [**Required**] The WOC shall improve Bandwidth Utilization across the circuit in which it is placed by a minimum of 30 percent.

TAC-001260 [**Required**] The WOC shall reduce latency across the circuit in which it is placed by a minimum of 30 percent.

A.8.3.3 Real-Time Traffic

TAC-001270 [**Required**] The WOC shall not degrade the transport efficiency of Real-Time Traffic below that of any un-optimized circuit in which it is placed.

A.8.3.4 Transparency

TAC-001280 [**Required**] The WOC shall not alter information content of either optimized or pass-thru traffic.

TAC-001290 [**Required**] Introduction of the WOC into the network infrastructure shall not disable any QoS Enforcement process, or exceed the performance parameters identified in Section 6, Network Infrastructure End-to-End Performance.

TAC-001300 [**Required**] The WOC shall not disable any Network Management process, or degrade the management functionality of any device identified in Section 2.17, Management of Network Appliances.

A.8.3.5 Routing

TAC-001310 [**Conditional**] If a WOC performs routing functions, then it shall provide a minimum of one of the following routing protocols:

a. Border Gateway Protocol (BGP) for inter-domain routing IAW RFCs 1772, 1997, 2385, 2439, 2544, 2796, 2918, 4271, and 4360.
b. Open Shortest Path First (OSPF), Version 2, for IPv4 and OSPF Version 3 for IPv6, July 2008, IAW RFC 5340.
c. Intermediate System-Intermediate System (IS-IS) for intra-domain routing IAW International Organization for Standardization (ISO)/International Electrotechnical Commission (IEC) 10589.

A.8.3.6 WOC Information Assurance

TAC-001320 [**Required**] The WOC shall comply with all applicable STIG requirements and policies, and shall meet Information Assurance requirements identified for Sections 11.2, DSN F-NE Generic, and 11.3, D-NE.

TAC-001330 [**Conditional**] If the WOC performs routing functions, then it shall meet the IA requirements identified for Router (R) within Section 4, Information Assurance.

A.8.3.7 WOC IPv6

TAC-001340 [**Required**] The WOC shall process IP version 6 (IPv6) traffic with performance equal to or better than IP version 4 (IPv4).

TAC-001350 [**Required**] The WOC shall meet the IPv6 requirements specified in Section 5, IPv6, identified for Network Appliance/Simple Server (NA/SS).

TAC-001360 [**Conditional**] If the WOC provides Layer 3 routing protocols, then the WOC must meet the IPv6 requirements specified for that protocol identified in Section 5, IPv6.

A.8.3.8 Scalability

TAC-001370 [**Required**] The WOC shall support a nominal expansion of services without requiring major overhaul or major replacement of equipment. The scalability requirements for WOC devices must be compliant with Section 3.4.4.2.4, Scalability.

A.8.3.9 Failover

TAC-001380 [**Required**] The WOC shall be resilient to loss of power. The product must self-restore to the last customer-configured state before the power loss, without intervention when power is restored. The WOC shall meet Section 7.6.6, Failover, (redundancy) requirements.

TAC-001390 [**Optional**] WOC devices should incorporate redundant power supplies.

TAC-001400 [**Required**] WOC devices shall default to “pass through” mode upon loss of system power. Pass through is defined as all ingress traffic must be successfully passed though the optimizer in an un-optimized format.